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USER'S MANUAL

Multi-Service IP PBX

IPBX 230

Version: 1.1
Date: 2006/11/07
P/N: 0700800096

TAINET IPBX 230 Administration Guidelines

Version 1.5 Beta 2 (build 0553)

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About This Manual

This section guides users on how to use the manual effectively. The manual contains information needed to install, configure, and operate TAINET's IPBX 230. The summary of this manual is as follows:

Chapter 1: Introduction

Presents overview, application, and other general information on the IPBX 230.

Chapter 2: System Configuration

This chapter describes how to configure system parameters used by TAINET IPBX 230.

Chapter 3: Service Configuration

This chapter describes details to configure various services built in the TAINET IPBX 230.

Chapter 4: IP PBX Configuration

This chapter introduces steps to provision the IP telephony part of the IP PBX.

Chapter 5: Feature Configuration

This chapter introduces a feature is a logical entity presenting a function module of IP PBX.

Chapter 6: Example Provisioning

This chapter introduces several practical configuration examples of TAINET IPBX 230 deployment.

Symbols Used in This Manual

3 types of symbols may be used throughout this manual. These symbols are used to advise the users when a special condition arises, such as a safety or operational hazard, or to present extra information to the users. These symbols are explained below:

Warning:

This symbol and associated text are used when death or injury to the user may result if operating instructions are not followed properly.

Caution:

This symbol and associated text are used when damages to the equipment or impact to the operation may result if operating instructions are not followed properly.

Note:

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Chapter 1. Overview

1.1 General Description

The TAINET IPBX 230 *Administration Guide* provides instructions for administering the TAINET IPBX 230 system. TAINET IPBX 230 is an embedded call-processing server communicating with client stations with Session Initiation Protocol (SIP). It migrates the telephony network and the data network of a small-to-medium business (SMB) company into a manageable converged network. TAINET IPBX 230 works with various IP phones (Desktop, WiFi, Bluetooth and DECT), voice-over-IP (VoIP) gateways, and PSTN network. Additional voice features such as conferencing, auto attendant, and voicemail are seamlessly enabled to all phones. TAINET IPBX 230 also provides Internet access to all LAN devices through Network Address Translation (NAT).

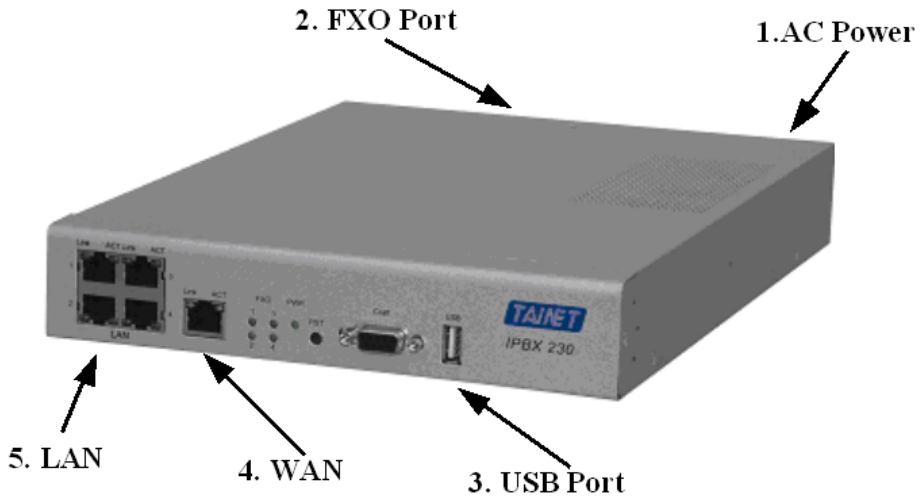
TAINET IPBX 230 provides call control and media relay services to SIP clients and applications. It performs the following primary functions:

- ACD (Automatic Call Distribution)
- Configurable multiple-layer IVR (auto attendant) with office-hour setting
- Configurable broadcast calls
- Fax relay and pass-through (T.38 and T.30)
- Voicemail IVR system
- Meet-me conference (Up to 8 parties per conference)
- SIP registrar (30 registrars)
- SIP outbound proxy for signaling and media
- SIP gateway (4 ports FXO)
- SIP PBX for extension calls

- Support codec (G.711u-law, G.711A-law, G.723.1 (high bit rate, low bit rate), G.729A, GSM, DVI4)

TAINET IPBX 230 has a built-in suite of voice applications for supplementary services, and no special-purpose hardware is required. Therefore, the total cost of ownership of a converged network enabled by TAINET IPBX 230 is lower than building separated infrastructures for legacy telephony network and data network. Moreover, it comes with a web-browsable interface to the data network configuration and voice service provisioning, which brings both local and remote manageability of networks together to facilitate administration.

1.2 Installation



1. AC Power	110/220 Volt, 60 Hz
2. POTS ports	Usually 4 ports per daughter card, numbered from right to the left. The rightmost port at slot 0 is port 1. FXO ports are to be connected to FXS jacks on wall or analog PBX using RJ-11 cables. FXO ports are to be connected to analog phones, or FXO ports of other devices.
3. USB ports	One external port with compliance to USB 1.1/2.0. Plug in a USB hard drive for CDR/voicemail backup from the internal storage.
4. WAN	Connect to a broadband modem or a WAN router.
5. LAN	Connect to a LAN switch.

Chapter 2. System Configuration

This section describes how to configure system parameters used by TAINET IPBX 230. The factory default of LAN IP address is 192.168.1.1. Connect to LAN port and the configuration Web interface is at **https://192.168.1.1/**. Once connected, the browser will ask for accepting a certificate. Click **Accept** to see the home page. Type in the default administrator ID and password (both are *admin*) to log in for administration. The administrator password can be changed in the **User Management -> User.**

Click **admin** in the **Login ID**.

Change the password in **Password**.

Click UPDATE to change the password.

Note:

For the system security, please change the password after the first login.

2.1 PBX System

The PBX System page briefs IP PBX status to the administrator. Firmware versions, IP addresses of WAN and LAN interfaces, and default gateway router are shown on this page. Click **PBX System** to see the basic information of IP PBX.

PBX Status	
Product Name :	IP PBX
Firmware AP Version :	1.5.0599
Firmware OS Version :	1.0.28(1)
WAN MAC Address :	00:90:BB:10:88:29
WAN IP Address :	172.16.5.150
WAN Subnet Mask :	255.255.255.0
Default Gateway :	172.16.5.254
LAN MAC Address :	00:90:BB:10:88:2A
LAN IP Address :	192.168.1.1
LAN Subnet Mask :	255.255.255.0

Figure 2-1 PBX Status

2.2 Time Setup

The Time Setup page allows administrator to configure time zone and date for TAINET IPBX 230. With correct time setup, functions such as IVR, worktime, and voicemail can present the actions at the right time. Select **System -> Time Setup**, and the current setting of time zone and date are displayed.

2.2.1 System Time Zone

Click a region/country in the Time Zone list, and click APPLY.

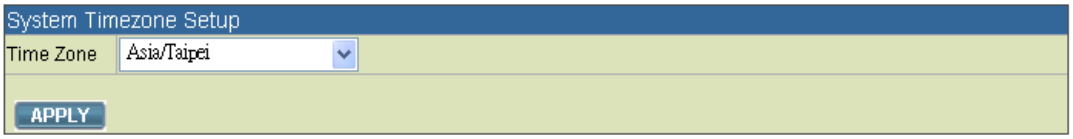


Figure 2-2 System Timezone Setup

2.2.2 Real Time Clock (RTC) Setup

Click year, month, day, hour, minute, and second in the correspondent list, and click APPLY.

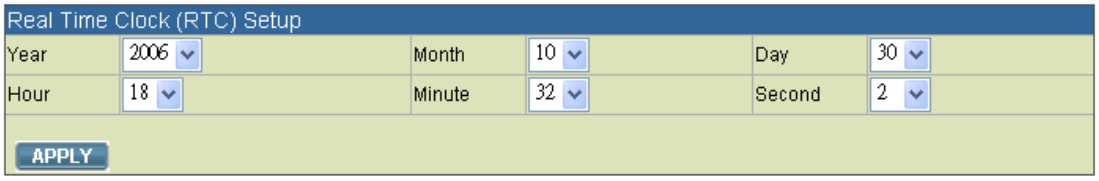


Figure 2-3 RTP Setup

Note:

When reset the time 15 minutes later than the time showed in RTC Setup, the system will ask for re-login.

2.3 On-board WAN Setup

The WAN Setup page allows administrator to configure WAN network interface for TAINET IPBX 230. Select **System -> On-board WAN Setup**, and the current setting of WAN network interface are displayed, e.g. type, IP address etc. Unless the **LAN Only** is selected, you can choose one of the three options, **Static IP**, **DHCP**, and **PPPoE** for your configuration. Select **LAN Only** check box to disable WAN, only default router and DNS settings are applicable.

2.3.1 Static IP

You can click **Static IP** in the **Type** list, and manually configure the following information:

- IP Address
- Netmask
- Default gateway IP address
- Primary, secondary or third DNS servers
- Click **APPLY** to submit.

2.3.2 DHCP

Simply click **DHCP** in the **Type** list, and click **APPLY**. The acquired IP Address, Netmask and Default Gateway information will show when revisit this page later.

2.3.3 PPPoE

- Click **PPPoE** in the **Type** list.
- Enter a user name and its password in **User Name** and **Password** boxes.
- Click **APPLY**.

The PPPoE dialing will start right away. When there is an active connection, the page will show the acquired IP address, network mask and default gateway information. Click **Disconnect** for not connecting.

2.3.4 LAN Only

Select **LAN Only** to disable WAN IP settings but allow the configuration of default gateway and primary/secondary/third DNS servers.

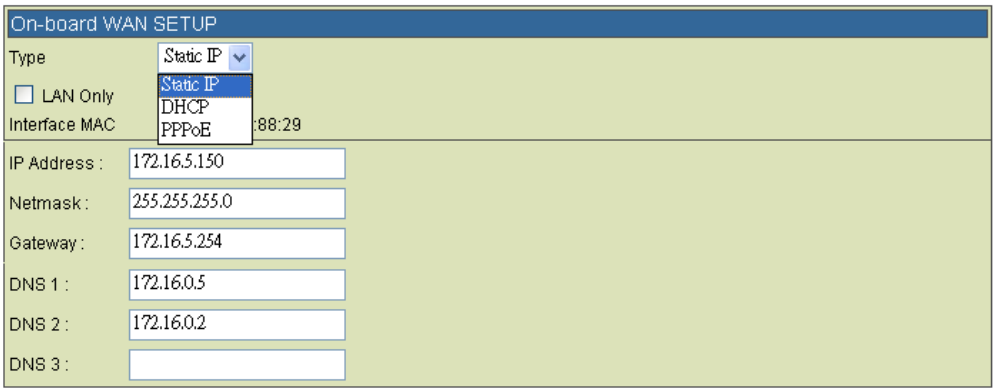


Figure 2-4 On-board WAN Setup

2.4 On-board LAN Setup

The LAN Setup page allows administrator to configure LAN network interface for TAINET IPBX 230.

- Select **System -> On-board LAN Setup** to see the current settings of LAN network interface.
- Enter a new IP address and network mask.
- Click **APPLY** to change the settings.

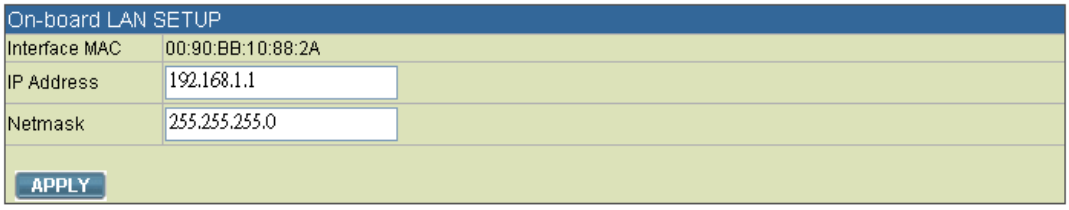


Figure 2-5 On-board LAN Setup

Note:

By default TAINET IPBX 230 grants IP addresses to LAN devices via DHCP, and translates those addresses into its WAN IP address for access beyond the LAN subnet. As a result, modifying the system LAN IP subnet must also change DHCP pool and LAN routing (if any) accordingly, and restart the IP PBX Service.

2.5 LAN Routing

To enable static routing among LAN subnets, enter network information and the IP address of the corresponding gateway in the IP PBX's LAN. It is important to assure that the given gateway IP address sits in the IP PBX's LAN. Each subnet requires an entry even multiple subnets share the same gateway, unless masking does the same. Examples are adding IP Route IDs *net1* and *net2* with parameters 192.168.128.0/255.255.255.0, 192.168.129.0/255.255.255.0, and shared gateway 192.168.1.254 respectively. Or, IP Route ID *net1n2* with 192.168.128.0/ 255.255.254.0 and gateway 192.168.1.254 would do the same. Added routes enable routing immediately after clicking **ADD**, however, the IP PBX Service needs to be restarted to regard calls from designated LAN subnets as LAN traffic.

2.5.1 Add a Route

- Enter the Route ID, Subnet, Netmask, and Gateway.
- Click ADD.

The newly added route will show in the **IP Route ID**.

2.5.2 Edit a Route

- Edit the information in a row.
- Click **APPLY** in the row to update the information.

2.5.3 Delete a Route

- Select a Route ID.
- Click **DEL**.

The deleted Route ID will disappear from the **IP Route ID** column.

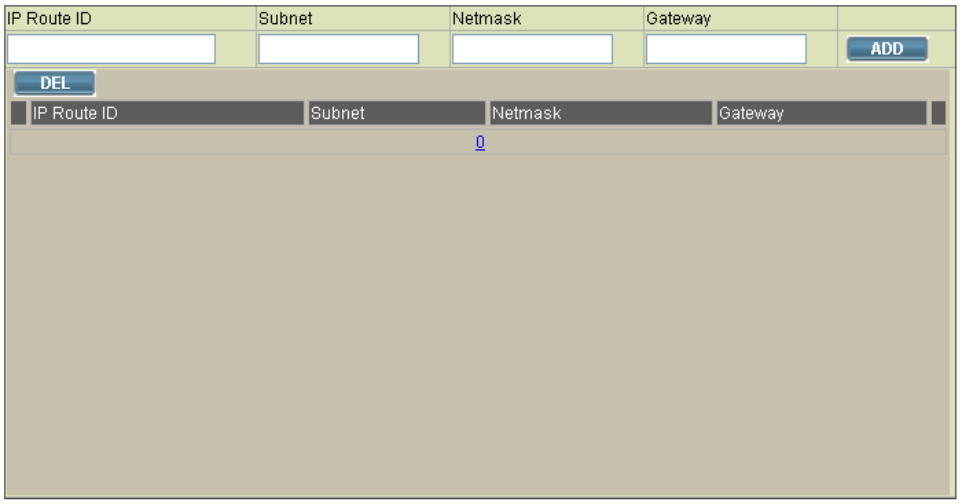


Figure 2-6 IP Route Table

Note:

When reset the time 15 minutes later than the time showed in RTC Setup, the system will ask for re-login.

2.6 Dynamic DNS Setup

Dynamic WAN IP address causes difficulty for inbound connections from remote clients or IP PBX systems. A popular work-around is to adopt domain names provided by Dynamic DNS service providers and run a client on or behind the gateway router (or IP PBX). It is required to apply an account and create a hostname in the account before configuration. Select **Enable** and give account info and refresh interval to activate a Dynamic DNS client. The client then uses **Username** and **Password** to access its account and update the **Hostname** with the latest WAN IP address at **DynDNS** or **3322.net Service** in **Interval** seconds periodically.

2.6.1 Enable Dynamic DNS

Typical hostname has a form of *<hostname>.dyndns.org* or *<hostname>.3322.net*. The refresh interval is usually between 60 – 600 seconds depending on the volatility of WAN IP assignment.

- Click **Enable**.
- Click **DynDNS** or **3322.net** in the **Service** list
- Enter the Username, Password, Hostname and Interval.
- Click **APPLY**.

2.6.2 Disable Dynamic DNS

Click **Disable**, and then click **APPLY**.

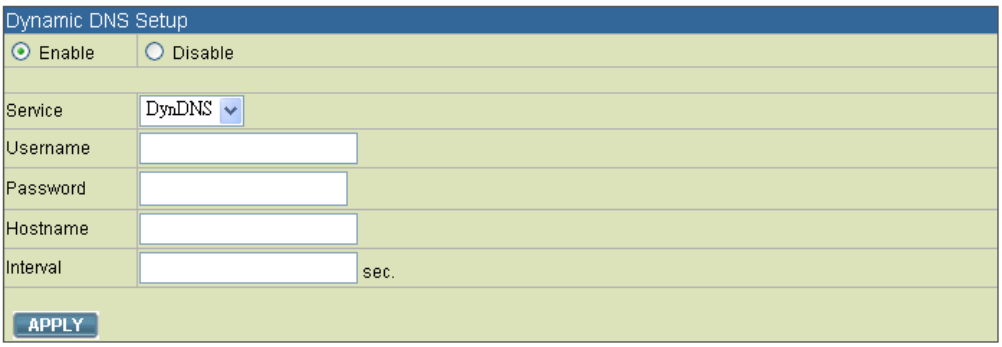


Figure 2-7 DDNS Setup

2.7 QoS Setup

To assure the bandwidth reserved for the outgoing VoIP traffic over regular data traffic from LAN, the QoS Setup page offers three parameters to characterize the WAN link. The default QoS setting is disabled because these parameters must be correctly given according to the actual WAN speed.

2.7.1 Enable QoS

Click **Enable** to enter the **WAN Uplink Speed**, **WAN Downlink Speed**, and **Uplink VoIP Reserved** (bandwidth), and click **APPLY**. For a popular 2M/256K ADSL program, the WAN uplink speed would be 256 and the WAN downlink speed would be 2048. The Uplink VoIP reserved could be, say, 192 out of the total 256 kbps to allow 2 concurrent G.711 calls.

2.7.2 Disable QoS

Click **Disable**, and then click **APPLY**.

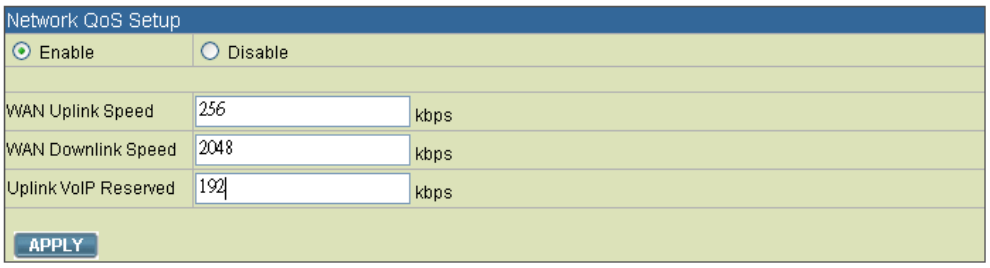


Figure 2-8 Network QoS Setup

2.8 Virtual Server

You can configure TAINET IPBX 230 as a virtual server for remote users to access services such as the Web or FTP at your local site via Public IP Addresses. With proper settings, TAINET IPBX 230 can automatically redirect s inbound traffic from WAN to local servers configured with private IP addresses. In other words, depending on the requested service (TCP/UDP) port number, the IP PBX redirects the external service request to the appropriate internal server (located at one of your LAN's Private IP Address). To enable access servers in LAN from a machine beyond WAN, select **System** -> **Virtual Server** to configure port mappings. **Service ID** names the service. **Protocol** and **Port** specify the TCP/UDP port number on WAN IP to be forwarded to the **Forward to Port** of **Forward to IP** in LAN. Say 192.168.1.5 is a Mail Server to be seen from outside, one should configure TCP port 25 to be forwarded to 192.168.1.5 port 25.

2.8.1 Add a Service

- Enter the Service ID, Protocol, Port, Forward to IP, and Forward to Port in correspondent boxes.
- Click **ADD**.

The latest Service will show in the **Service ID**.

2.8.2 Edit a Service

- Change any information in a row.
- Click **APPLY** in the row to update the information.

2.8.3 Delete a Service

- Select a **Service ID**.
- Click **DEL**.

The deleted **Service** information will disappear from the **Service ID**.

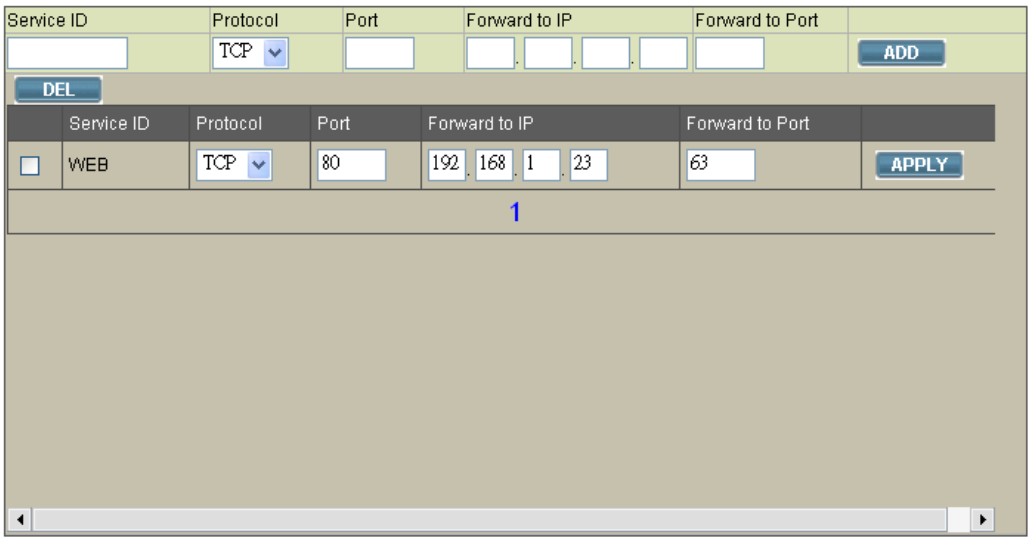


Figure 2-9 Virtual Server Setup

2.9 Maintenance

This page includes maintenance functions of IP PBX, including **Storage Backup**, **SIP UA**, **CDR Log**, **System Event** and **Active Calls**.

2.9.1 Storage Backup

To back up internal main storage, click **BACKUP**, and follow the instructions to insert the USB connector of an external USB drive. Options include whether to keep or remove CDR and/or voicemails after backup. After a confirmation of the insertion, backup starts a few seconds later if the external USB drive is accessible and has enough free space. If the backup is successful, a new folder will be created on the external drive. After the backup, remove the USB connector of the external drive.

2.9.2 SIP UA

SIP UA Status lists the registration status of each client and remote IP PBX and the **IP Address/Port** from where they register. SIP trunk registrations, if any, also show at the end of the list. The **Dynamic** column shows the listed IP address is dynamic or static. **Reg. Progress** is the response code and message if registration has been attempted but not successful so far. **Slave Registrar** column is used only under the stackable mode. It indicates with which slave box a SIP client is registered. Blank means a client is registered with the master box locally.

2.9.3 CDR Log

The CDR (Call Detail Record) Log shows each call record including Calling and Dialed Numbers, Caller ID, Destination Interface (trunk if outbound) in use, epochs when the call was made, answered, and ended, which yield the total and billable durations. The last column denotes the disposition of a call like answered or not.

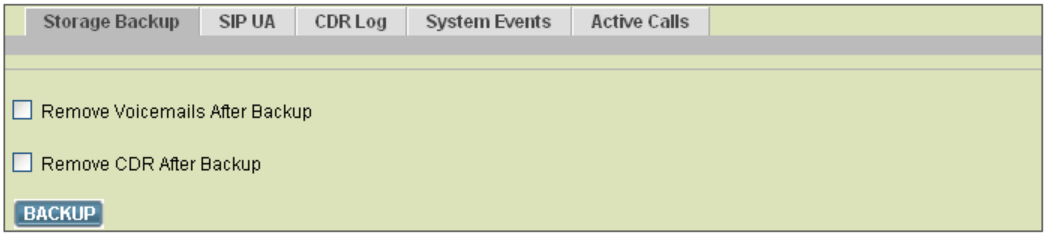
And you can get all the CDR log through “get file” button, the file format is Excel file.

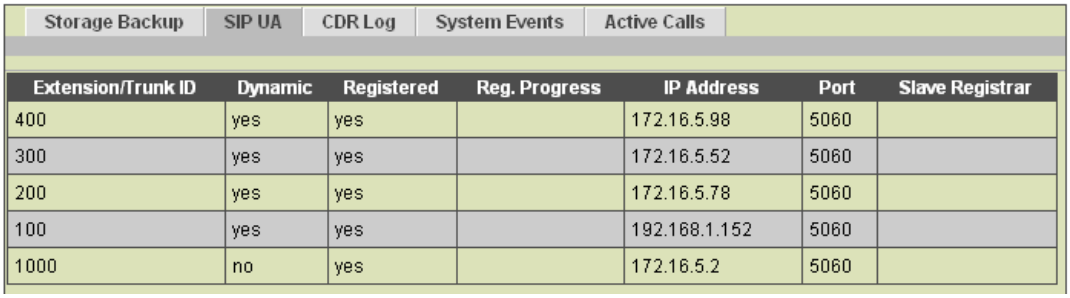
2.9.4 System Events

Event log includes reported events from following system services: NTP, DNS, DHCP and PPPoE.

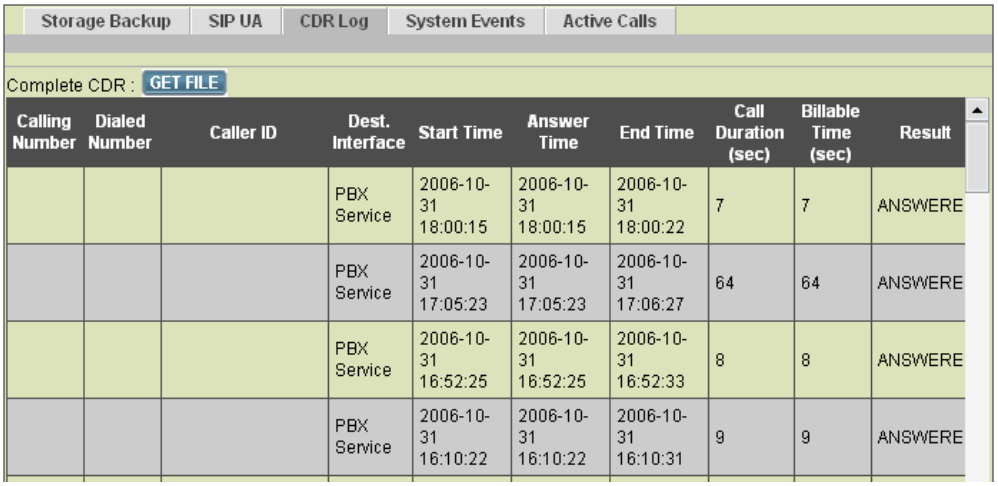
2.9.5 Active Calls

The Active Calls page shows current active calls. Columns Client and Party indicate the involved extensions or trunks of a call. State shows the state of a call, while Service gives the current action of the listed Client.





Extension/Trunk ID	Dynamic	Registered	Reg. Progress	IP Address	Port	Slave Registrar
400	yes	yes		172.16.5.98	5060	
300	yes	yes		172.16.5.52	5060	
200	yes	yes		172.16.5.78	5060	
100	yes	yes		192.168.1.152	5060	
1000	no	yes		172.16.5.2	5060	



Calling Number	Dialed Number	Caller ID	Dest. Interface	Start Time	Answer Time	End Time	Call Duration (sec)	Billable Time (sec)	Result
			PBX Service	2006-10-31 18:00:15	2006-10-31 18:00:15	2006-10-31 18:00:22	7	7	ANSWERE
			PBX Service	2006-10-31 17:05:23	2006-10-31 17:05:23	2006-10-31 17:06:27	64	64	ANSWERE
			PBX Service	2006-10-31 16:52:25	2006-10-31 16:52:25	2006-10-31 16:52:33	8	8	ANSWERE
			PBX Service	2006-10-31 16:10:22	2006-10-31 16:10:22	2006-10-31 16:10:31	9	9	ANSWERE

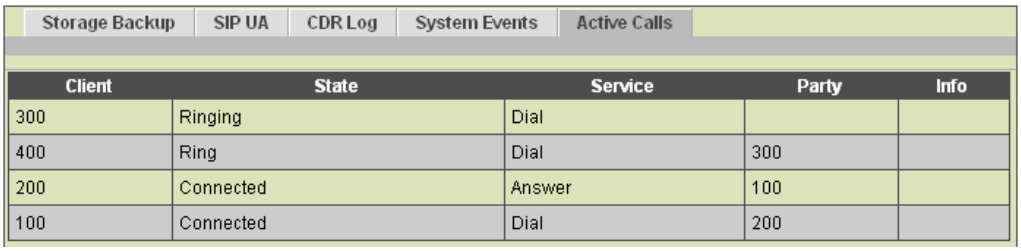


Figure 2-10 Maintenance

Table 2.9-1 Maintenance Description

Field	Description	
Client	Show the caller or callee's extension number, port number, or SIP trunk ID.	
State	Connected	In the conversation.
	Ring	The client is a caller and is ringing a callee.
	Ringing	The client is a callee and is ringed by a caller.
	Reserved	FXS detects off-hook.
Service	Dial	The client is a caller.
	Answer	The client is a callee.
	IVR	Calls from FXO are picked up by Auto-Attendant.
	Meet-me	The client enters meet-me.
	Voicemail	The client enters voicemail.
Party	Shows extension number, POTS number or SIP trunk ID that is talking to this client.	

2.10 Firmware Upgrade

The version of the running PBX firmware could be found in **System** -> **Firmware Upgrade**, incluse the Application firmware version and OS firmware version. To upgrade current firmware, click **Browse** to locate a release file obtained from the vendor, and click **UPGRADE** to have the latest version of PBX firmware.

Figure 2-11 Firmware Upgrade

Warning:

Do not change the firmware file name; otherwise the system will reject it.

2.11 Reboot

By selecting **System -> Reboot**, you can reboot the machine by clicking **YES**. In case the software reboot fails, you can also press the hardware front panel **Reset** button. It is advised to shut down IP PBX system before a power-off.

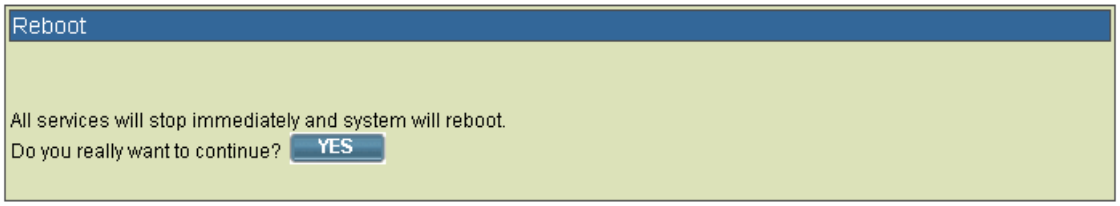


Figure 2-12 Reboot

Chapter 3. Service Configuration

This section describes details to configure various services built in the TAINET IPBX 230.

3.1 NTP Service

Select **Service -> NTP Service** to specify a NTP server for network time synchronization. You can enable or disable NTP service at any time.

3.1.1 Enable NTP Service

- Click **Enable**.
- Select **Automatic** check box to use server pool at pool.ntp.org; or, enter a fully qualified domain name or the IP address of a NTP server.
- Click **APPLY**.

3.1.2 Disable NTP Service

Click **Disable**, and click **APPLY**.

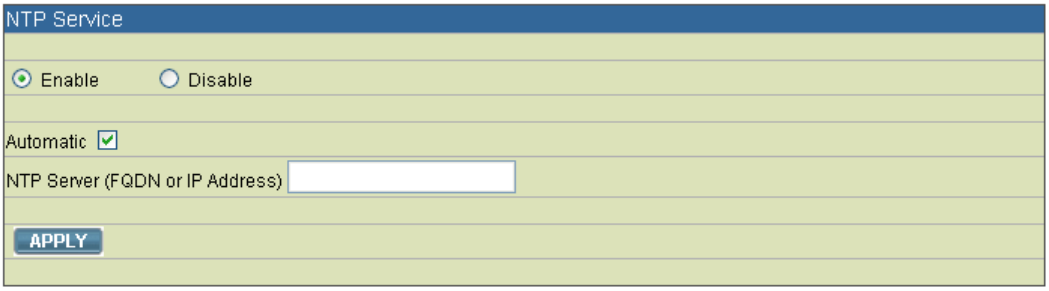


Figure 3-1 NTP Service

3.2 STUN Service

TAINET IPBX 230 has a built-in STUN client to solve NAT problems. Select **Service -> STUN Service** to specify a Simple Traversal of UDP through NATs (STUN) server for NAT traversal. You can enable or disable STUN Service at any time.

Note:

You have to restart the IP PBX Service, after changing the STUN setting.

3.2.1 Enable STUN Service

- Click **Enable**.
- Enter a fully qualified domain name or the IP address of a STUN server.
- Click **APPLY**.
- Go to **Service -> IP PBX Service**, and click **RESTART** to reflect the changes.

3.2.2 Disable STUN Service

Click **Disable**, enter the fully qualified domain name or the static IP address of the external WAN interface and then click **APPLY**. Usually this address refers to the static WAN IP address if there is a NAT device between the IP PBX and the Internet. If the WAN port of IP PBX directly connects to Internet or it is unused, leave the address blank. Go to **Service -> IP PBX Service**, and click **RESTART** to reflect the changes.

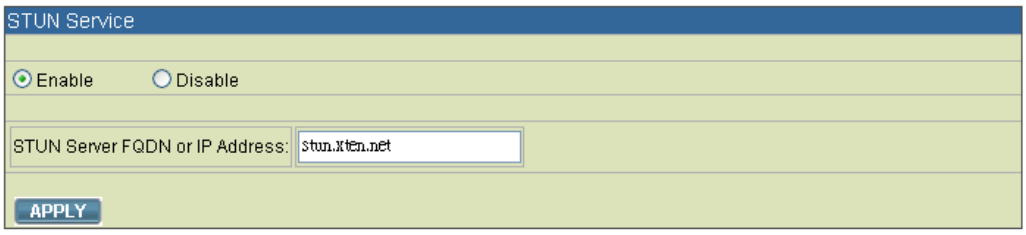


Figure 3-2 STUN Service

3.3 TFTP Service

Select **Service -> TFTP Service** to view the current status of TFTP Service. You can enable or disable TFTP Service at any time.

3.3.1 Enable TFTP Service

Click Enable, and then click APPLY to manage files, e.g. upload and download files to and from the IP PBX. Uploaded files can then be retrieved through TFTP Service.

3.3.1.1 Change Directory

Current directory is shown in the gray field on the right side of Directory, for instance, it is /. at the beginning. Click a directory in the Directory list to change to a different folder.

Note:

The default directory is /. Initially, you may not be able to change the directory, since no folder is created under / yet.

3.3.1.2 Add a Folder

- Click a directory under which you want to add a new folder in the **Directory** list.
- Click ADD FOLDER.
- Enter a folder name in the pop-up dialog box, e.g. myfolder.
- Click **OK**.

The new folder is then created accordingly, e.g. /myfolder/.

3.3.1.3 Delete a Folder

- Click a directory of a folder in the **Directory** list.
- Click DELETE FOLDER.

The deleted folder shall disappear from the **Directory** list.

Note:

A folder cannot be deleted if there is still file inside.

3.3.1.4 Download a File

- Click a directory in the **Directory** list.
- Click a file in the Download / Delete File from the Above Folder list.
- Click **GET FILE** to download the file.

3.3.1.5 Delete a File

- Click a directory in the Directory list.
- Select a file in the Download / Delete File from the Above Folder list.
- Click DEL FILE to delete the file.

3.3.1.6 Upload a File

- Click a directory in the Directory list.
- Click Browse.
- Select a directory in the Find list, and then a file.
- Click Open.
- Click PUT FILE to upload the file.

Now, the uploaded file should appear in current directory and is displayed in the **Download / Delete File from the Above Folder** list.

3.3.2 Disable TFTP Service

Click Disable, and then APPLY.

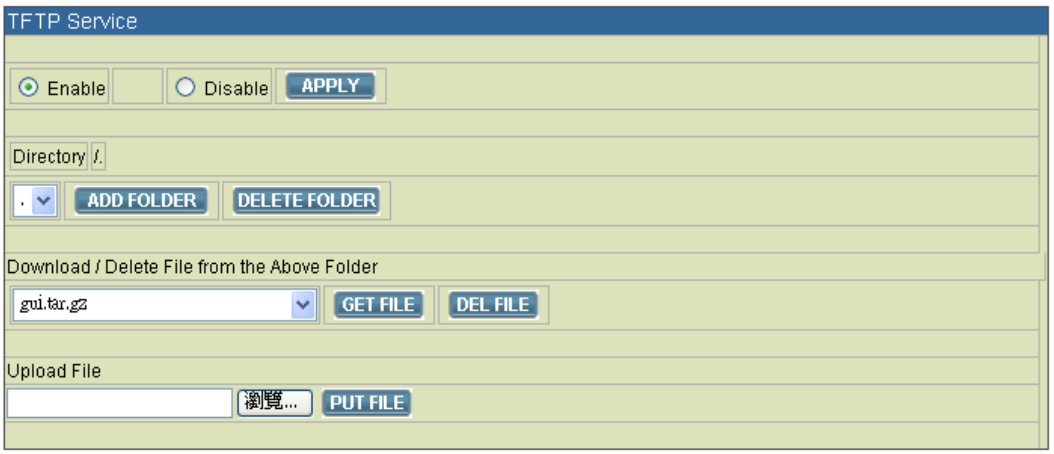


Figure 3-3 TFTP Service

3.4 DHCP Service

Select Service -> DHCP Service to view the current status of the DHCP Service. You can enable or disable the DHCP Service at any time.

Note:

If the IP PBX was shut down abnormally, Select Service -> DHCP Service and click APPLY, or Go to Service -> IP PBX Service, and click RESTART to active the DHCP service.

3.4.1 Enable DHCP Service

Click Enable, choose the main interface offering addresses, and then APPLY to configure DHCP settings.

3.4.1.1 Add DHCP Range

- Click CLEAR.
- Enter a pool name (must have an alphabet initial) in Pool Name.
- Select Single-host to enter an IP address of the host with MAC, if the binding is intended for a specific host only.
- Enter a DHCP range of addresses available for lease in IP. The right address box will not show if Single-host is selected.
- Optionally, DHCP options1 could be configured by entering an option code and value in Code, Value and click ADD. The new DHCP option will show in the OPTIONS list. To delete an option, choose it from the OPTIONS list and click DEL after the box.
- Click ADD at the bottom of the page to commit changes.

You can see the newly added DHCP POOL displayed in the **DHCP POOL** list.

3.4.1.2 Edit DHCP Range

- Click any pool name in the **DHCP POOL** list to see the settings on the right.
- Edit the settings.
- Click **UPDATE** to change the settings.

3.4.1.3 Delete DHCP Range

- Click any pool name in the **DHCP POOL** list.
- Click **DEL**.

The deleted pool name will disappear from the **DHCP POOL** list.

3.4.1.4 Show Clients

Click SHOW CLIENTS to see all leased LAN IP addresses and client details.

3.4.2 Disable DHCP Service

Click Disable, and click APPLY.

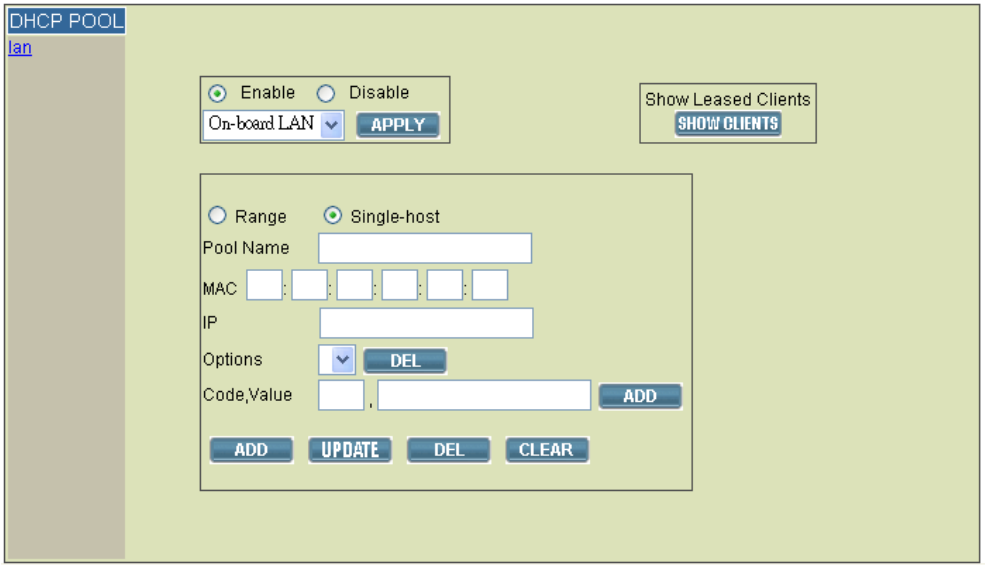


Figure 3-4 DHCP Service

3.5 IP PBX Service

In Service -> IP PBX Service, you can click the Service & Configuration tab to reload, backup, restore, restart or revert the IP PBX configuration, or click the Advance tab for the IP PBX parameters settings.

3.5.1 Service & Configuration

Select Service-> IP PBX Service, and then click the Service & Configuration tab.

3.5.1.1 Reload IP PBX Configuration

Click RELOAD, and IP PBX will reload the configuration once there is no active call. If there is any active call, it will retain up to 3 minutes, and then IP PBX will reload. This is the most frequently used function in this page since any IP PBX configuration change has to be reloaded to take effect.

3.5.1.2 Backup IP PBX Configuration

Click BACKUP, and IP PBX archives and encrypts current configuration into a time-stamped backup file under tftpboot root directory. To secure configuration files, download them to a local host through the GET FILE function in Service -> TFTP Service once a while. Clear PBX Settings Only check box, both PBX and system (interfaces and services) settings will be archived in the backup file.

Warning:

Do not change the configuration file name, or the RESTORE function will reject the configuration file.

3.5.1.3 Restore IP PBX Configuration

Click a configuration backup file in the list, click **RESTORE**, and IP PBX will restore the configuration as current setup. Go to **Service** -> **IP PBX Service**, and click **RESTART** to activate the settings.

3.5.1.4 Restart

Click RESTART, and the IP PBX Service will restart completely. Currently active calls will be disconnected immediately. This function is rarely required unless the network setting has been changed, or the service operates abnormally without problematic configuration could be identified.

3.5.1.5 Revert

Click REVERT, and IP PBX will erase current IP PBX settings and revert configuration back to the factory default. Note the reversion affects IP PBX service only, but not other system services such as DHCP, TFTP, and NTP. The backup IP PBX configuration files under TFTP remain intact after reversion, so that one can restore to a specific time if a backup file had been generated then.

To revert the whole system back to the factory default as much as possible, hold the hardware Rest button for 10 seconds. Since this will wipe out almost everything generated by the user, all system interfaces and services must be configured from scratch again if no appropriate backup configuration could be restored. Note that such reversion will not erase backup configurations and existing voicemails. Backup configuration files could be deleted in the TFTP Service page and voicemails could be deleted in the Maintenance page.

3.5.2 Advance

Select Service-> IP PBX Service, and then click the Advance tab to configure IP PBX parameters.

Table 3.5-1 IP PBX Service – Advance Settings

Field	Description
PBX SIP Port	Specify the UDP port where the SIP service listens on.
RTP Port Range	Limit the UDP ports used by the IP PBX for media transport. ☞ The port range needs to have at least equals to the (number of extensions (also count shared-lines) + number of SIP trunks (also count trunk terminals)) * 2. If selecting **Enable Video Codec**, the total amount needs to multiply by 2 to have the least requirements

Field	Description
	for RTP port range.
Max/Default Expiration Time	Guard and advertise SIP registration respectively.
PBX Caller ID	The default Caller ID for an unknown incoming call.
Enable Video Codec	Select if there will be video clients registering to the system
Max Active Users	Enter a number for registration admission control to limit the maximum number of active registered clients.
Max Active Calls	Enter a number for call admission control to limit the maximum number of concurrent calls.
Max Wireless Calls	Enter a number to limit the calls made by explicitly specified wireless extensions.
IP TOS Value	Set the TOS value in the IP header of RTP packets originated from IP PBX.
Disable WAN Bandwidth Saver	Select to disable attempts to use low-bit-rate codec (G.729A or G.723.1) for remote parties.

Chapter 4. IP PBX Configuration

This section introduces steps to provision the IP telephony part of the IP PBX. Please note that reloading configuration is required in order to make new configuration effective.

4.1 User Configuration

A user is a logical entity in IP telephony, which associates extensions with a usergroup. It also propagates its attributes such as e-mail and voicemail PIN to extensions. Usually a user refers to a real person who has a name and e-mail; however, one can always create virtual users to associate with public extensions. For example, extensions in reception, break room and lab areas.

The User Management page allows the administrator to manage users in the IP telephony network. Select User Management-> User, and one can add, edit, and delete users. Go to Service -> IP PBX Service, and click RELOAD to activate changes.

4.1.1 Add a User

- Click **ADD**.
- Enter settings shown in Table 4.1-1.
- Click **ADD**.
- Click **BACK** to see the newly added user in the **Login ID**.

4.1.2 Edit a User

- Click a user in the Login ID.
- Edit settings shown in Table 4.1-1.
- Click UPDATE.

4.1.3 Delete a User

- Select a Login ID.
- Click DEL.

The deleted user shall disappear from the **Login ID** column.

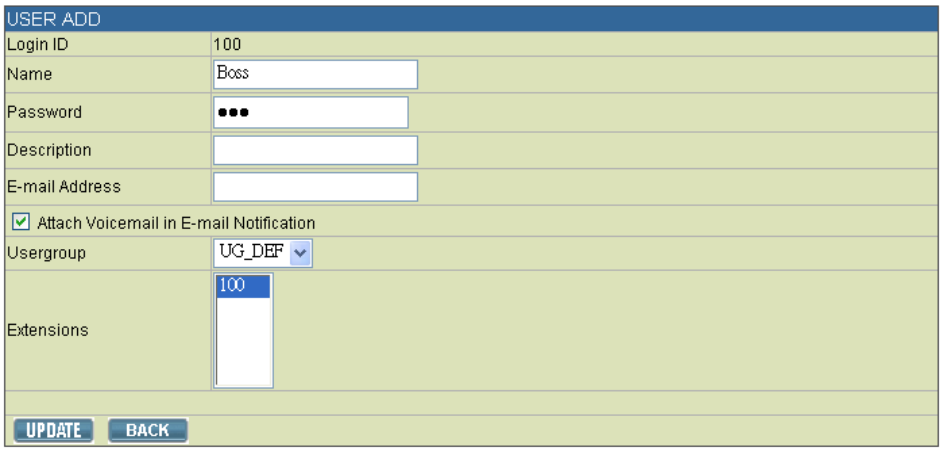


Figure 4-1 User Add

Table 4.1-1 User configuration Settings

Field	Description
Login ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum. This is the ID for personal configuration through IP PBX Web management.
Name	Name of the user, either a real or a virtual one, e.g. Alice Lee or Conference Room.
Password	Password for the user to access IP PBX Web management.
Description	Arbitrary description information.
E-mail Address	E-mail address of the user for voicemail notification.
Attach Voicemail in E-mail Notification	Select to enclose the message received in the notification e-mail as an attachment.
Usergroup	Select the usergroup this user belongs to. ☞ There is no usergroup initially. Come back later to revise this selection if no appropriate usergroup could be chosen for now.
Extensions	Show the extensions associated with this user.

4.2 User Group Configuration

A usergroup is a logically grouping of users and their privileges. For instance, one could have couple of usergroups in an IP telephony network, e.g. Sales, Marketing, Administration, Accounting, and Engineering, etc. Each usergroup associates with a set of PBX features and call routing scopes. In other words, all users in the same usergroup share the same reachability of PBX features and final destinations.

The User Group Management page allows the administrator to manage usergroups. Select **User Management-> User Group**, and one can add, edit, or delete usergroups. Go to **Service -> IP PBX Service**, and click **RELOAD** to activate changes.

4.2.1 Add a User Group

- Enter a usergroup name beside the ADD button, and then click ADD.
- The name will show in Group ID.
- Click the name in Group ID to view the edit page.
- Enter settings shown in Table 4.2-1.
- Click SET to save the settings, and click BACK to return to the USERGROUP MANAGEMENT page.

Now, you can see the newly added usergroup displayed in the Group ID.

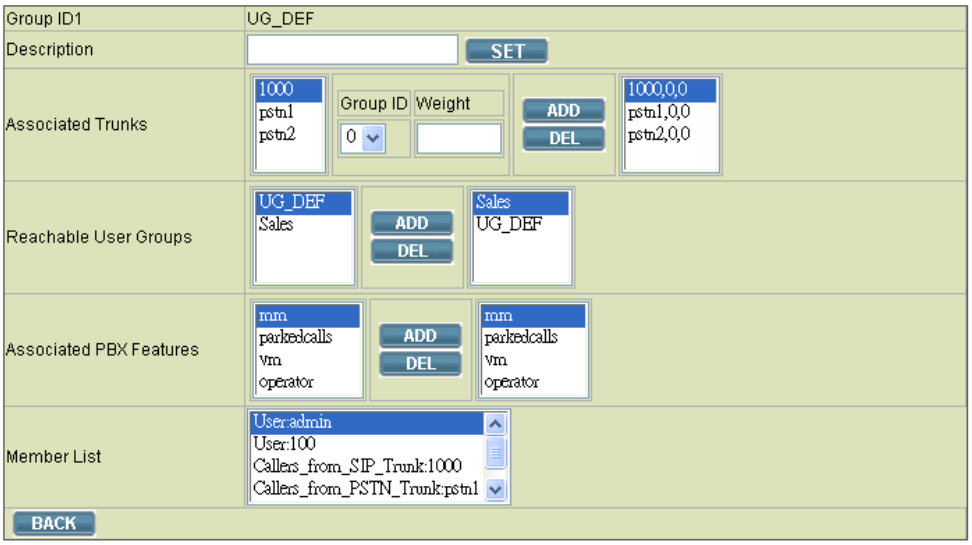
4.2.2 Edit a User Group

- Click a usergroup name in the **Group ID**.
- Edit settings shown in Table 4.2-1
- Click **SET**.

4.2.3 Delete a User Group

- Select a **Group ID**.
- Click **DEL**.

The deleted usergroup shall disappear from the Group ID column.



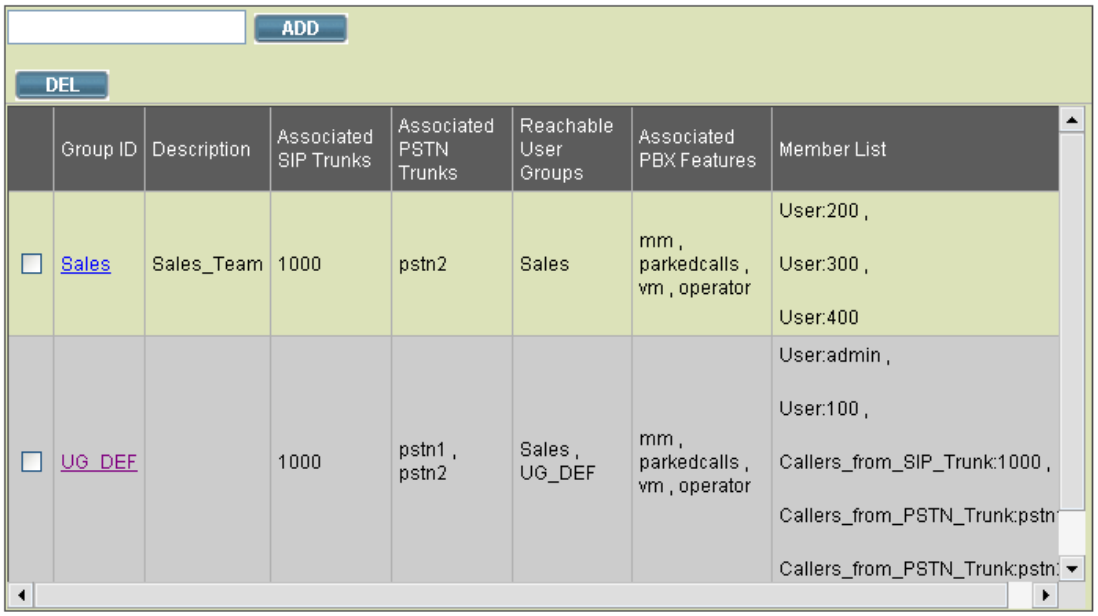


Figure 4-2 User Group

Table 4.2-1 Usergroup Configuration Settings

Field	Description
Group ID	A unique group name containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description information.
Associated Trunks[2]	Select outbound SIP trunks and PSTN trunks accessible by this usergroup. Note the order matters the hunting sequence in run-time. *Group ID*: The default number is "0". A trunk with Group ID "0" does not form a balance group with any other trunks in Group 0. If Group ID is 1~9, trunks with the same Group ID form a usage balance group. *Weight*: the weight of a trunk to be selected in a trunk balance group for an outgoing call. *Note*: Trunks with the same group ID must be put together, or the function will not work. ☞ **If there is no SIP trunk and PSTN trunks initially, come back later to revise selection once trunks have been created.**
Reachable User Groups	Select other usergroups reachable from this usergroup. By default, only users in the same usergroup can be reached one another. ☞ If there is no usergroup initially, come back later to revise this selection, once more usergroups have been created.
Associated PBX Features[3]	Select PBX features enabled to this usergroup. Here vm stands for Voice Mail, mm for Meet-me Conference, parked calls for Call Parking. ☞ Most features have to be configured to function correctly. Remember to examine the settings of selected features before activating current configuration.
Member List	Show the users associated with this usergroup. ☞ If there is no user initially, come back later to select, once one or more users have been created and associated with this usergroup.

[2] Please refer to 4.6, 4.7 for details.
[3] Please refer to 0 for details.

4.3 Device Configuration

A device could be an IP phone, gateway, analog telephone adapter, or even another IP PBX, etc. It has one or more extensions to be registered to the IP PBX.

4.3.1 IP Phone

The DEVICE PHONE MANAGEMENT page lets the administrator to create IP Phone devices. Before a device can be reached from the IP PBX, the same account information has to be programmed into the device through the configuration interface enabled by the device. Select Device -> IP Phone, to add, edit, and delete devices. Go to Service -> IP PBX Service, and click RELOAD to activate changes.

4.3.1.1 Add a Device

- Enter a device name in the **Device ID** box, and a URL in the **Device Administration URL** box.
- Click **ADD**.

The newly added device will display in the Device ID.

4.3.1.2 Edit a Device

Once create the device, you can modify its information through the following steps.

- Modify the **Device Administration URL** and click **LINK** as a shortcut to the device administration URL.
- Click **EDIT** to see the Enable Automatic Client Configuration (ACC) page. is a reference for detailed ACC settings which is used for auto-configuring IP phones, one can specify the MAC address and audio preferences of the phone. Note that for phones using HTTP for auto-configuring, DHCP setting needs a new option 151 with a value of *http://<IP PBX LAN IP>/tftpboot/* in the **Code, Value** box in **Service -> DHCP Service**. No extra settings needed if the phone uses TFTP for auto-configuring.

- Click **ENABLE** to see **Enable** shows in the A**uto Client Conf** column. Click **EDIT** and then **DISABLE** to disable the function.

4.3.1.3 Delete a Device

- Select a **Device ID**.
- Click **DEL**.

The deleted device shall disappear from the Device ID.

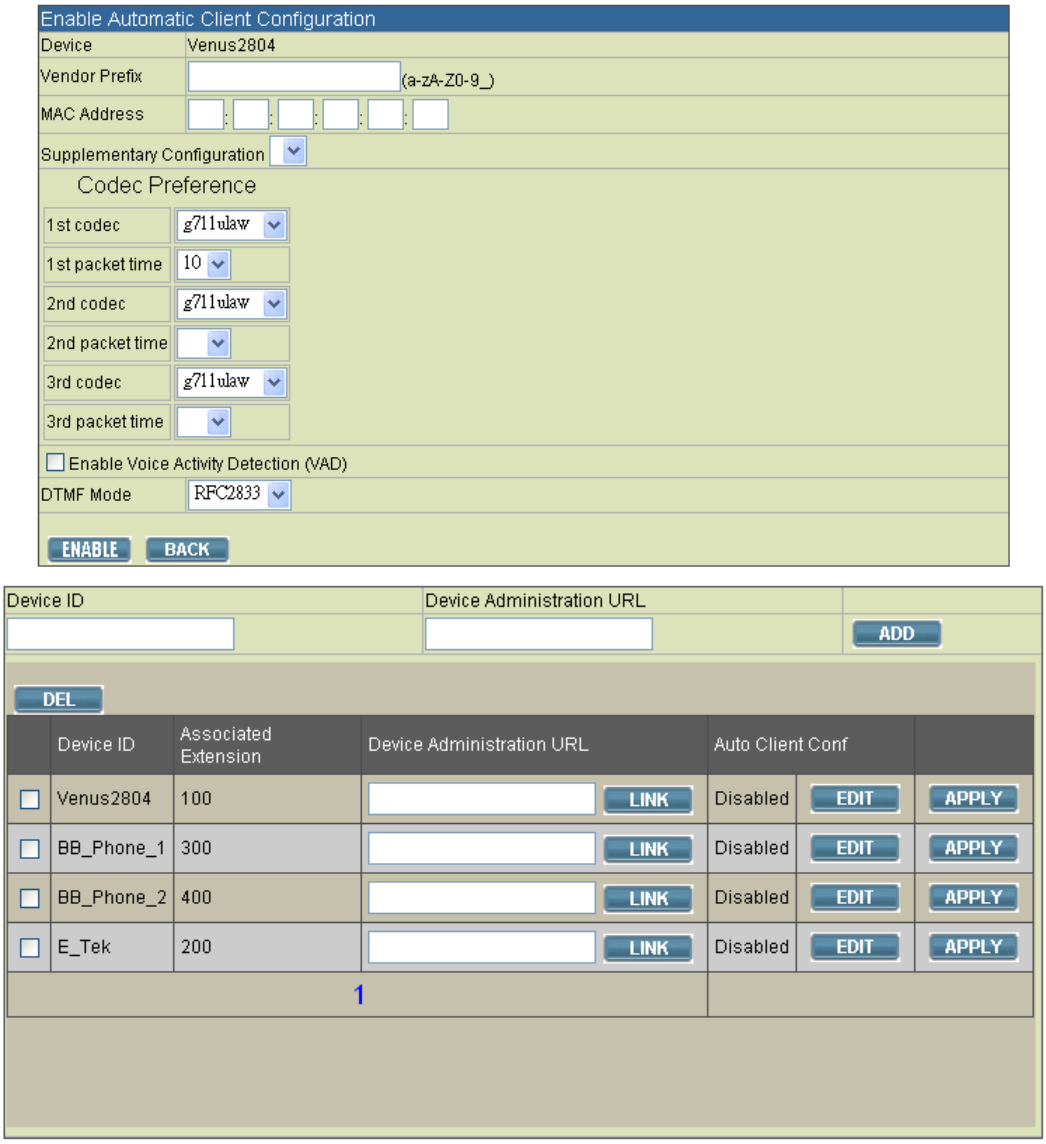


Figure 4-3 ACC Configuration

Table 4.3-1 ACC (Automatic Client Configuration) Settings

Field	Description
Device	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Vendor Prefix	Ask your IP Phone vendor for the Prefix.
MAC Address	MAC address of the device.
Supplementary Configuration	Specify if provided by the phone.
Codec Preference	Preference order of supported codecs and packet times of the phone.
Enable Voice Activity Detection (VAD)	VAD is a technique that detects absence of audio and conserves bandwidth by preventing the transmission of "silent packets" over the network. ☞ Select if your IP Phone supports VAD.
DTMF mode	Choose a DTMF mode used by the phone.

4.3.2 Extension of IP Phone

The EXTENSION MANAGEMENT page lets the administrator to create extensions. Select Device -> Extension of IP Phone, and one can add, edit, and delete extensions. Go to Service -> IP PBX Service, and click RELOAD to activate changes. TAINET IPBX 230 also offers its own extensions Line-in-use Call Back function. For example, Extension A and B are within one IP PBX or a stacked cluster. When extension A calls extension B, and extension B is in use. Extension A can press 1 to be in the waiting list. Once extension B finishes the call, IP PBX will ring extension B within a minute. When extension B is picked up, IP PBX will ring extension A to connect the call. Extension A call also press * to cancel, and will connect to voice mail.

4.3.2.1 Add an Extension

- Click **ADD** to set an extension.
- Enter settings shown in Table 4.3-2.
- Click **ADD**.
- Click **BACK** to see the newly added extension.

4.3.2.2 Edit an Extension

- Click an extension in the **Extension Number**.
- Edit settings shown in Table 4.3-2.
- Click **UPDATE**.
- Click **BACK** to see the updated information.

4.3.2.3 Delete an Extension

- Select an extension numbers.
- Click **DEL**.

The deleted extension shall disappear from the **Extension Number**.

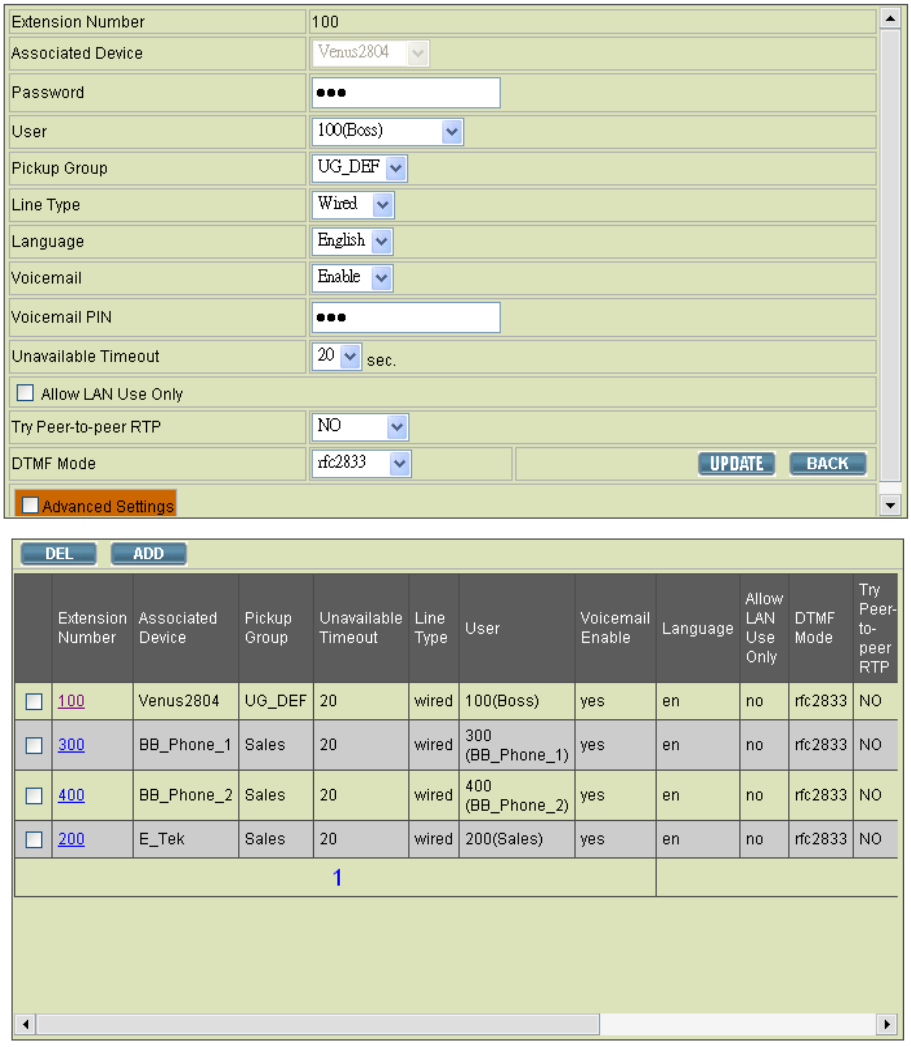


Figure 4-4 Extension IP Phone

Table 4.3-2 Device Extension Configuration Settings

Field	Description
Extension Number	A unique line number composed of digits only, e.g. 101; 32 digits maximum. This is the login ID on the device configuration side.
Associated Device	Select the Device this extension associates with.
Password	Password of this extension. Same password must be configured on the device side as well.
User[4]	Select the user this extension associates with. ☞ There may not be appropriate users to select initially. One can come back later, once the expected user has been added.
Pickup Group	The pickup group the extension belongs to. Extensions in the same pickup group can call *8 to pick up a call in ringing state.
Line Type	Specify the type of connection, wired or wireless, of the client with the extension.
Language	Preferred language for system instructions heard from the extension.
Voicemail	Select enable to allocate voicemail account for the extension.
Voicemail PIN	PIN to access voicemails. This is mandatory if above voicemail option is enabled.
Unavailable Timeout	Timeout for ringing before a call is answered.
Allow LAN Use Only	Check to reject registration and calls from WAN in a SIP ID same as the extension number. I.e., this extension must be on LAN.
Try Peer-to-peer RTP	If click YES, IP PBX will attempt to notify the two peers in a conversation to try peer-to-peer RTP transmission. This is suggested as long as phones support INVITE or UPDATE method during a connected call to save the resource of IP PBX. However, only SIP INFO DTMF mode phones should enable this since other DTMF modes require IP PBX being

[4] Please refer to 4.1 for details.

Field	Description
	RTP relay server to support in-line transfer.
DTMF Mode	Choose preferred DTMF mode for this extension. Currently supported types include RFC2833, SIP INFO, and in-band tone. It must match configuration on the device side.
Advanced Settings	Select to see more optional settings that is shown below.
Selective Call Blocking	(Optional) Select Block Anonymous Calls to block all calls without a Caller ID.
	(Optional) Block one or more calling numbers by entering the calling numbers and clicking ADD. Removing the blocked numbers by clicking the number from the list, and then click DEL.
Forward Options	(Optional) Select Unconditional Call Forward and clicks a default destination in the list, e.g. Voicemail or Phone Number. ☞ If selecting Phone Number, enter a number to which incoming calls are forwarded unconditionally. The number could be an extension or a PSTN number with appropriate outbound prefix.
Unavailable Call Forward	(Optional) Enter a number to which incoming calls are forwarded when not answered. The number could be an extension or a PSTN number with appropriate outbound prefix.
Timeout To Next Forward	(Optional) Timeout before trying next forwarding number in the list. Note that if the forwarded number has personal setting of forwarding policy, this timeout guards the total duration allowed before the personal setting connects a call. As long as the call does not go through, eventually it returns to the hunt list of forwarding.
Play Unavailable Forward Prompt	(Optional) Notify the caller that callee is not available and the call is being forwarded to another extension.
Line In Use Forward	(Optional) Enter a number to which incoming calls are forwarded when the extension is busy. The number could be an extension or a PSTN number with appropriate outbound prefix. ☞ If the function is enabled, the busy call back function will be disabled.

Field	Description
Selective Call Forward	(Optional) Unconditional Call Forwarding according to the calling number. Enters one or more calling numbers and a forwarding number, and clicks ADD. E.g., forward only calls from 101 to a cellular number, while let the rest enter the voice mail by default. Selects a forwarding and click DEL when the forwarding is no longer required.

4.4 Route Configuration

A route is a destination number pattern for outbound call matching. A pattern consists of digits 0-9 (including “-”), “*”, “#”, digit set, and wildcard characters like “.”, “X”, “Z”, and “N”. explains digit set and wildcard characters.

Table 4.4-1 Digit Set and Wildcard Characters for Route Patterns

Expression	Description
[<digits>]	Match any single digit listed explicitly. E.g., digit set [13579] match odd digits. One may use ‘-‘ to indicate a range of digits, e.g. [2-8].
.(dot)	Match any digit in any length. Usually given in the end of a pattern to include all numbers matched a specific prefix. ☞ .(dot) can not be used alone or at the beginning of the route patterns.
X	Match any single digit from 0 to 9.
Z	Match any single digit from 1 to 9.
N	Match any single digit from 2 to 9.

By selecting Route Management-> Route, the administrator can add, edit, and delete routes in the Route Management page. Go to Service -> IP PBX Service, and click RELOAD to activate changes.

4.4.1 Add a Route

- Enter settings shown in Table 4.4-2.
- Click **ADD**.

The newly added route should be displayed in the Route ID.

4.4.2 Edit a Route

- Edit settings shown in 4.4-2 in a row.
- Click **APPLY** in the row to update the settings.

4.4.3 Delete a Route

- Select a **Route ID**.
- Click **DEL**.

The deleted route shall disappear from the Route ID.

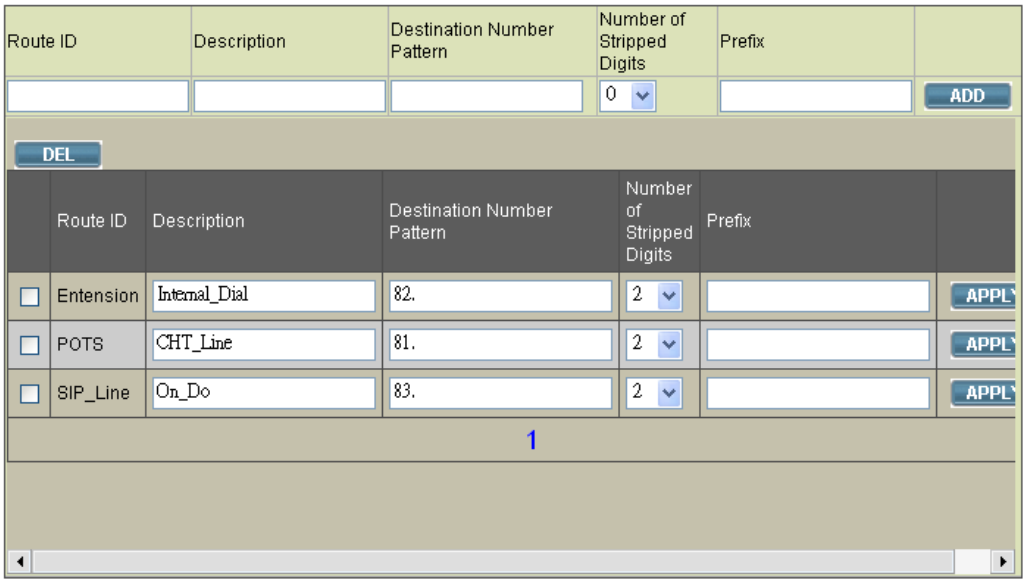


Figure 4-5 Route Setup

Table 4.4-2 Route Configuration Settings

Field	Description
Route ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description information.
Destination Number Pattern	A destination number pattern consisting of digits, digit set, and wildcard characters, e.g. 9NXXXXXX matches any 7-digit called number starting from a digit larger or equal to 2 and with an extra prefix digit 9.
Number of Stripped Digits	Number of leading digits to be stripped from the original dialed number when matches this route. Using 9NXXXXXX as an example route pattern with number of stripped digits equal to 1, dialing 95270001 will be stripped to be 5270001 when it actually got dialed out.
Prefix	A sequence of digits to be prefixed to the final dialed number after stripping. Using 9NXXXXXX as an example

Field	Description
	route pattern with number of stripped digits equal to 1 and prefix 1408; dialing 95270001 will be 14085270001 when it actually got dialed out. A special prefix character “w” could be used for PSTN trunks to pause 0.5 second during dialing. Say, 4 leading consecutive “w” result in 2 seconds delay before dialing.

4.5 Route Group Configuration

A routegroup groups routes into a logical superset of route patterns. Such abbreviation simplifies the association of multiple routes with a trunk, say and a PSTN line. A route must be included into at least one routegroup in order to take the route pattern into effect.

By select Route Management-> Route Group, the administrator can add, edit and delete routegroups in the ROUTE GROUP MANAGEMENT page. Go to Service -> IP PBX Service, and click RELOAD to activate changes.

4.5.1 Add a Route Group

- Type a route group name and click **ADD**.
- Click the route group in **Group ID** to see the settings.
- Enter settings shown in and click **BACK**.

The newly added route group should be displayed in the Group ID.

4.5.2 Edit a Route Group

- Click a route group name in Group ID.
- Edit settings shown in Table 4.5-1 .
- Click SET, if there is any update in the Description box.
- Click BACK to see the updated information.

4.5.3 Delete a Route Group

- Select a Group ID.
- Click DEL.

The deleted route group shall disappear from the Group ID.

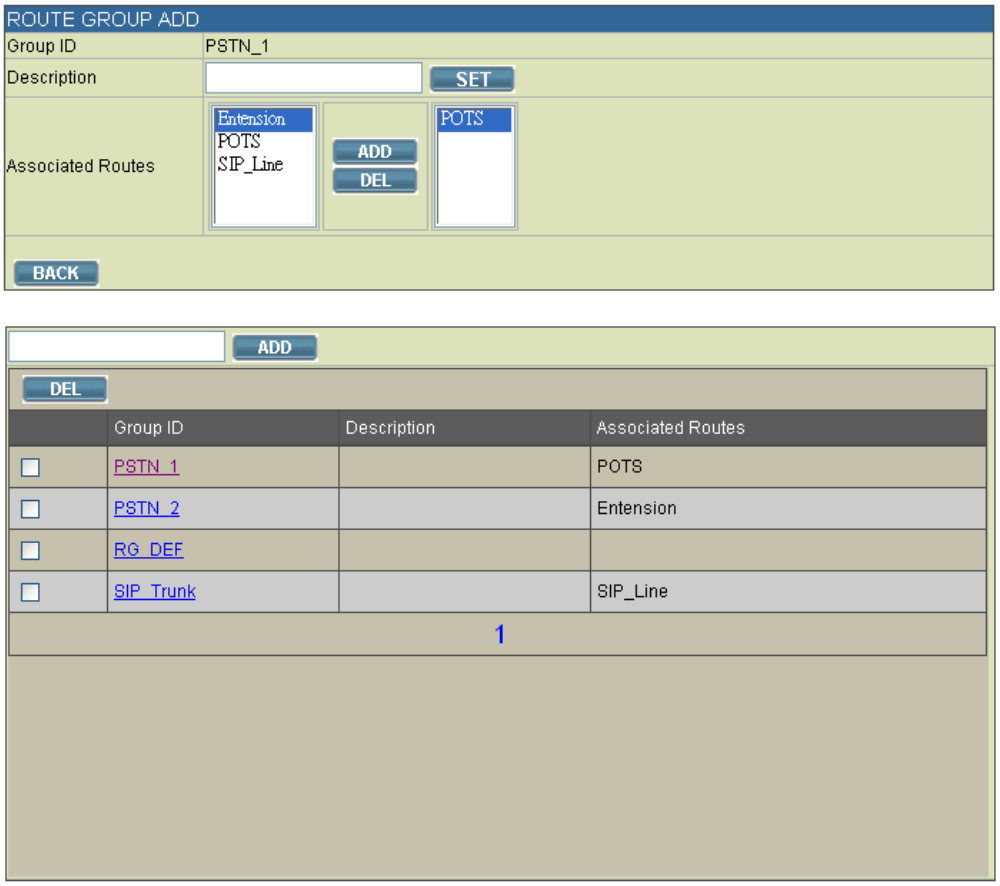


Figure 4-6 Route Group Setup

Table 4.5-1 Routegroup Configuration Settings

Field	Description
Group ID	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Description	Arbitrary description information.
Associated Routes[5]	Select routes belonged to this routegroup. Click ADD/DEL button to add or remove a route to or from the routegroup. The right box lists current selected routes. Note the order of the selected routes is important since it decides which route would be matched first for an outgoing call. ☞ **There may not be appropriate routes to select initially. One can come back later to revise it, once the expected routes are added.**

4.6 SIP Trunk Configuration

A SIP trunk refers to a SIP account on a remote call routing or gateway device. A practical example is an account at an Internet Telephony Service Provider (ITSP) where a call is routed to a SIP client or off-ramped to an analog subscriber via PSTN. One could also build SIP trunk to a remote IP PBX to reach its extensions and PSTN ports.

The SIP TRUNK MANAGEMENT page allows the administrator to configure SIP trunks used by TAINET IPBX 230. Select Trunk -> SIP Trunk, and one can add, edit, and delete SIP trunks. Go to Service -> IP PBX Service, and click RELOAD to activate changes.

4.6.1 Add a SIP Trunk

- Click the **Add New** tab.
- Enter settings shown in Table 4.6-1.
- Click **ADD**.

The newly added SIP trunk shall be displayed in Trunk Identifier.

4.6.2 Edit a SIP Trunk

- Click the Trunks tab, and More to see more information.
- Edit settings shown in 4.6-2 in a row.
- Click APPLY in the row to update the information.

4.6.3 Delete a SIP Trunk

- Click the Trucks tab, and select a trunk identifier.
- Click DEL.

The deleted SIP trunk shall disappear from the Trunk Identifier.

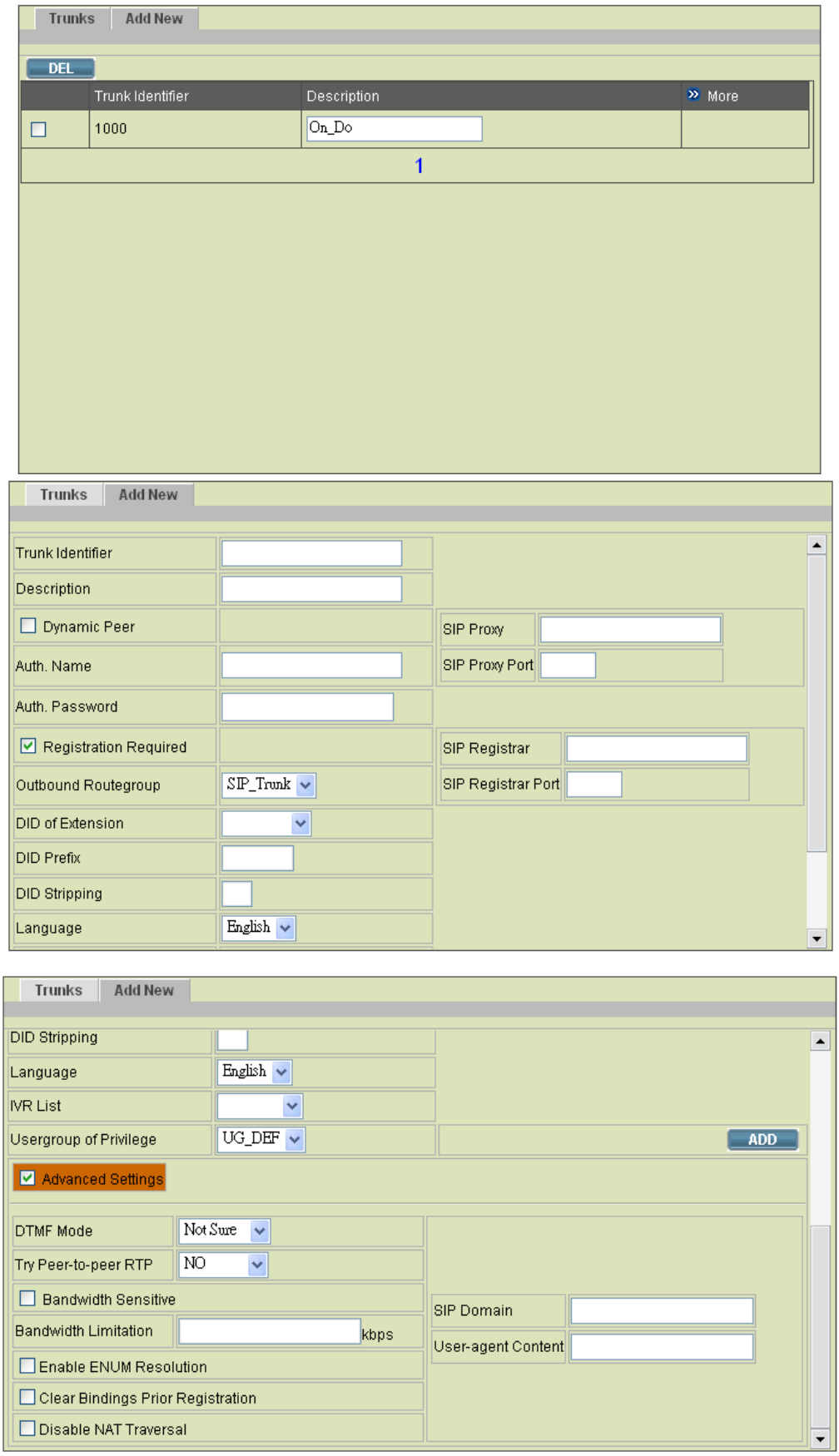


Figure 4-7 SIP Trunk Setup

Table 4.6-1 SIP Trunk Configuration Settings

Field	Description
Trunk Identifier	A unique number consisting of digits only. Usually give the phone number issued by the ITSP for consistency.
Description	Arbitrary description information.
Dynamic Peer	Select if the trunk is a passive trunk which means the registration will be from a dynamic remote peer. Typical application is to accept registration from an IP PBX at a remote site with dynamic IP address. Once the remote IP PBX registers, calls from local to remote can be made reversely over the trunk.
SIP Proxy SIP Proxy Port	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP proxy, which usually refer to the SIP server on the ITSP side.
Auth. Name	Specify the name for authentication if different to the Trunk ID.
Auth. Password	Give the password used for authentication on the remote SIP proxy or registrar. Usually this is given by the ITSP.
Registration Required	Select if registration to a registrar is required to activate the trunk. This is true for a remote IP PBX or an ITSP account, however, may be not required in case of a SIP gateway.
SIP Registrar SIP Registrar Port	Specify IP address (or fully qualified domain name) and UDP port of the remote SIP registrar, which usually refer to the SIP server on the ITSP side (same as proxy).
Outbound Routegroup[6]	Select a routegroup to associate routes with this trunk. Outbound calls match included route patterns could employ this trunk to hop onto a remote SIP domain. ☞ There may not be any appropriate routegroup to select initially. One can come back later to revise it, once the expected routegroup has been added.
DID of Extension	When enabled DID, clicks an extension in the list to be an unconditional destination for incoming calls to this trunk. Or click bynumber and then enter configurations in DID

[6] Please refer to 0 for details.

Field	Description
	Prefix and DID Stripping to have the incoming calls directed to the corresponding extension derived by number manipulation. The SIP trunk numbers is therefore regarded as the direct line of the extension. ☞ If you set a DID extension in a trunk, then only that extension can use this trunk to call out, and all incoming calls to this trunk will connect to that extension directly.
DID Prefix	A digit string to be prefixed to the incoming called number after stripping.
DID Stripping	A number of leading digits to be stripped from the original called number. If prefix or stripping has been given but DID of Extension is not bynumber, the result of digit manipulation is dialed in a DTMF string after the call has been answered by the DID extension as an automatic 2nd dialing.
Language	Preferred language for system instructions heard from the trunk.
IVR List[7]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID.
Usergroup[8] of Privilege	When disabled DID, click a usergroup in the list whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this trunk. ☞ There may not be appropriate usergroups to select initially. One can come back later once the expected usergroup has been added.
Advanced Settings	Select to see more settings that is shown below.
DTMF Mode	Select a preferred DTMF mode, RFC 2833 or SIP INFO, for this trunk in the list. This must match configuration on the server side. If the user does not know the DTMF mode on the server side, select Not sure from the list, and SDP will automatically detect the DTMF mode is Inband or RFC2833.
Try Peer-to-peer	Click NO to disable or IP PBX will attempt to notify the

[7] Please refer to 5.12 for details.
[8] Please refer to 4.2 for details.

Field	Description
RTP	two peers in a conversation to try peer-to-peer RTP transmission. This is suggested as long as phone and ITSP side support INVITE or UPDATE method during a connected call to save the resource of IP PBX. However, only SIP INFO DTMF mode should enable this since other DTMF modes require IP PBX being RTP relay server to support in-line transfer.
Bandwidth Sensitive	Indicate the trunk is over a bandwidth-sensitive link, e.g. across Internet.
Bandwidth Limitation	Leave it blank to disable or, specifies a limit of bandwidth in kbps for call admission.
SIP Domain	Specify the SIP domain used by the proxy and registrar. If not specified, IP address will be used as the domain by default.
User-agent Content	Override default User-Agent header content.
Enable ENUM Resolution	Select to use ENUM resolution, or leave it as blank.
Clear Bindings Prior Registration	Select if failed to the registration, and cannot identify any abnormal settings.
Disable NAT Traversal	IP PBX uses NAT traversal for outgoing traffics by default. Select to disable NAT traversal if there is a machine that could handle NAT issues.

4.7 FXO PSTN Trunk configuration

A FXO PSTN trunk group is a logical group of one or more FXO PSTN subscriber lines connecting to FXO ports on TAINET IPBX 230.

The FXO PSTN TRUNK MANAGEMENT page allows the administrator to configure PSTN trunks. Selec Trunk -> FXO PSTN Trunk, and one can add, edit and delete PSTN trunks. Go to Service -> IP PBX Service, and click RELOAD to activate changes.

4.7.1 Add a FXO PSTN Trunk

- Click the Add New tab.
- Enter settings shown in Table 4.7-1.
- Click ADD.

The newly added PSTN Trunk shall display in the Trunk Group.

4.7.2 Edit a FXO PSTN Trunk

- Click the Trunks tab, and click More to see more information..
- Enter settings shown in a row of the trunk group.
- Click APPLY in the row to update the information.

4.7.3 Delete a FXO PSTN Trunk

- Click the Trunks tab, and select a trunk group.
- Click DEL.

The deleted PSTN trunk shall disappear from the Trunk Group.

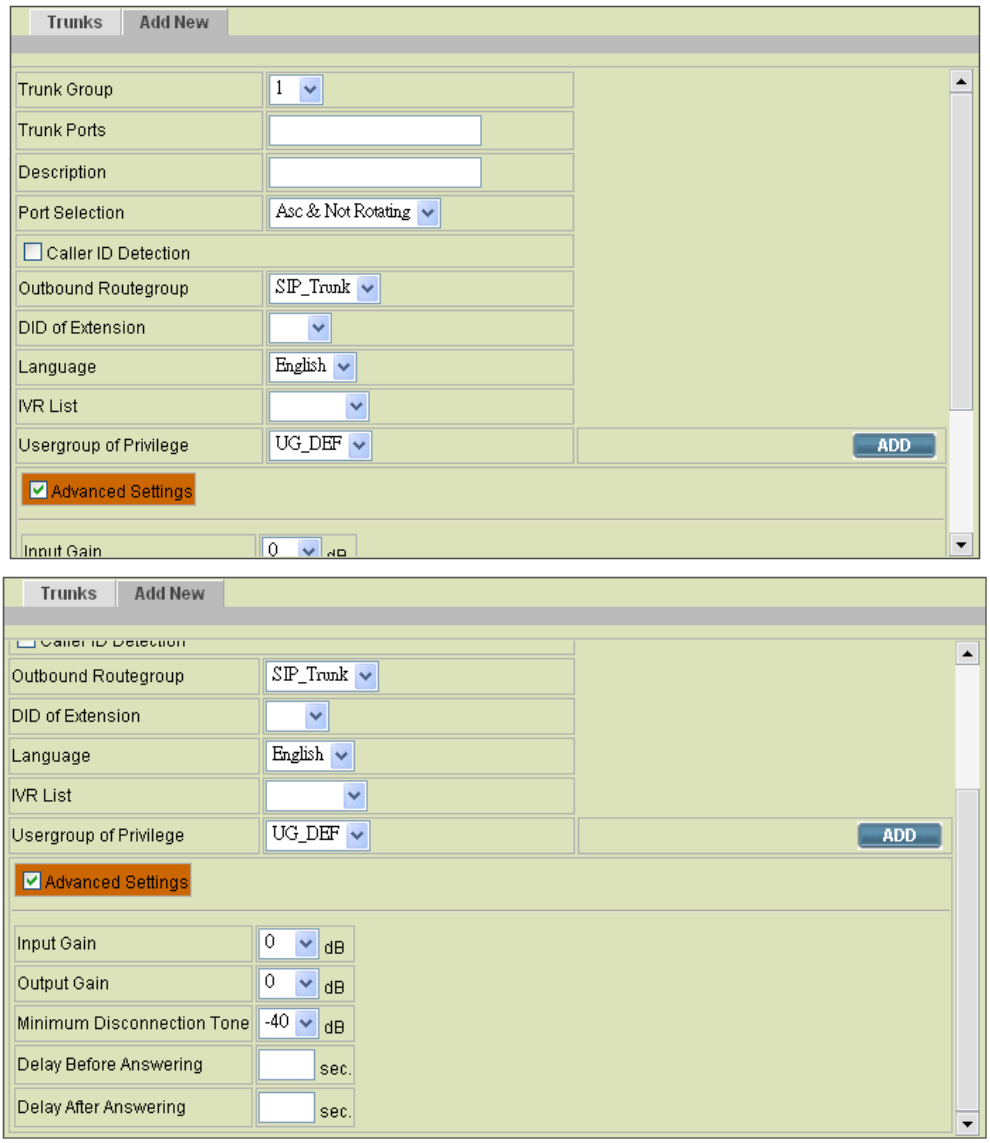


Figure 4-8 FXO PSTN Trunk Configuration

Table 4.7-1 FXO PSTN Trunk Configuration Settings

Field	Description
Trunk Group	ID number of this PSTN trunk group. A valid number ranges from 1 to 31.
Trunk Ports	FXO port indices grouped by this PSTN trunk, such as 1 or 1,2 or 1-3, etc. Maximum port index depends on the number of physical ports available.
Description	Arbitrary description information.
Port Selection	Click to search for an available port in the group. Asc mean is Ascending, Dsc mean is Descending. Rotating means to force ports being selected by turns to even cost.
Caller ID Detection	Select to detect the Caller ID calling from PSTN lines.

Field	Description
Outbound Routegroup[9]	Select a routegroup to associate routes with this trunk. Outbound calls match included route patterns could employ this trunk to access PSTN. ☞ There may not be appropriate routegroup to select initially. One can come back later to revise it, once the expected routegroup is added.
DID of Extension	When enabled DID, clicks an extension in the list to be an unconditional destination for incoming calls to this trunk. The PSTN numbers of the included ports are therefore regarded as the direct line numbers of the extension. ☞ If you set a DID extension in trunk, then only that extension can use this trunk to call out, and all other user's call in this trunk will connect to that extension.
Language	Preferred language for system instructions heard from the trunk.
IVR List[10]	Associate an IVR menu with incoming calls to this trunk. This is mandatory unless the trunk is configured for DID.
Usergroup[11] of Privilege	When disabled DID, click a usergroup in the list whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this trunk. ☞ There may not be any appropriate usergroups to select initially. One can come back later to revise it, once the expected usergroups are added.
Advanced Settings	Select to see more settings shown below.
Input/Output Gain	Voice amplification or attenuation in dB scale to adjust input/output volume of a PSTN line.
Minimum Disconnection Tone	Minimum volume level of the disconnection tone. If a PSTN trunk is found to have disconnection problem and voice sounds low, choose a lower dB.
Delay Before/After Answering	Delay in seconds before and after answering a call from PSTN trunk.

[9] Please refer to 0 for details.
[10] Please refer to 5.12 for details.
[11] Please refer to 4.2 for details.

4.8 Terminal Trunk Configuration

A SIP trunk terminal refers to a SIP account for a remote SIP trunk to register with. It terminates SIP registration and invitation from a remote IP PBX and relay calls to local clients, PSTN trunks, or further SIP trunks. In a site-to-site SIP trunking application, a SIP trunk on one side usually pairs with a trunk terminal on the other side to form a unidirectional call hand-off path. To allow trunking in the other direction, the two sides swap roles and form another pair. Since a trunk terminal is the account for a SIP trunk to authenticate with, exact the same identifier and password must be used for both.

The TERMINAL TRUNK MANAGEMENT page allows the administrator to configure trunk terminals used by TAINET IPBX 230. Select Trunk -> Terminal Trunk, and one can add, edit and delete terminals. Go to Service -> IP PBX Service, and click RELOAD to activate changes.

4.8.1 Add a Terminal Trunk

- Click the Add New tab.
- Enter settings shown in Table 4.8-1.
- Click ADD.

The newly added terminal shall display in the Terminal Identifier.

4.8.2 Edit a Terminal Trunk

- Click the Trunks tab, and More to see more information.
- Edit settings shown in 4.8-1 in a row.
- Click APPLY in the row to update the information.

4.8.3 Delete a Terminal Trunk

- Click the Trunks tab, and select a terminal identifier.
- Click DEL.

The deleted terminal trunk shall disappear from the Terminal Identifier.

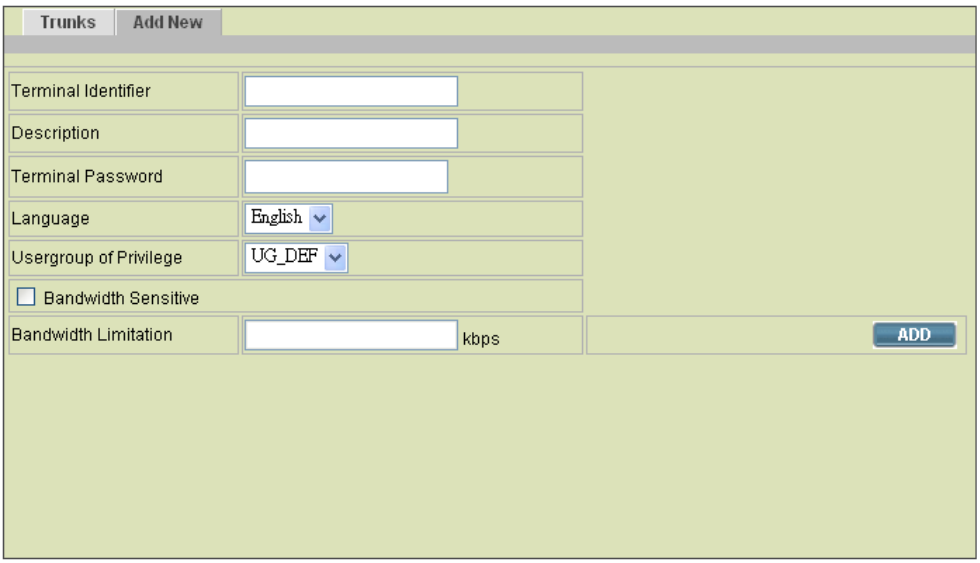


Figure 4-9 Trunk Terminal Setup

Table 4.8-1 Trunk Terminal Configuration Settings

Field	Description
Terminal Identifier	A unique number consisting of digits only. This is the trunk identifier configured on the other IP PBX.
Description	Arbitrary description information.
Terminal Password	Password of SIP trunk given on the other IP PBX for authentication.
Language	Preferred language for system instructions heard from the terminal.
Usergroup[12] of Privilege	When disabled DID, click a usergroup in the list whose reachability to other usergroups and trunks will be used as the privilege of inbound calls from this terminal. ☞ There may not be any appropriate usergroups to select initially. One can come back later, once the expected usergroup has been added.
Bandwidth Sensitive	Indicate the trunk is over a bandwidth-sensitive link, e.g. across Internet.
Bandwidth Limitation	Leaves this blank to disable or, specifies a limit of bandwidth in kbps for call admission.

[12] Please refer to 4.2 for details.

4.9 POTS Setting

This page allows selection of country-based progress tones and/or impedance and/or compact type of POTS ports. Click APPLY to save modifications. Go to Service -> IP PBX Service, and click RESTART to active new settings.

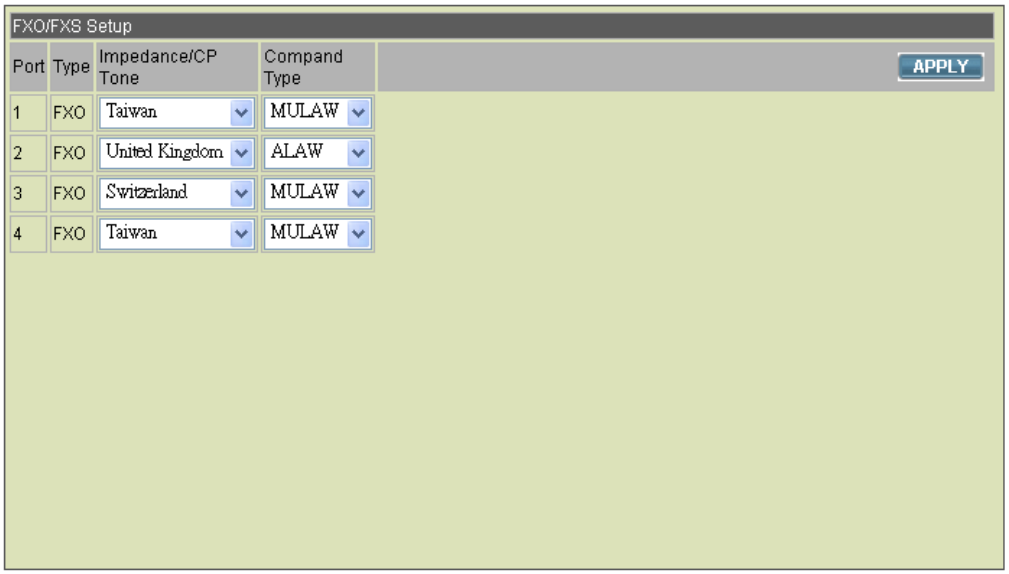


Figure 4-10 POTS Setting

Chapter 5. Feature Configuration

A feature is a logical entity presenting a function module of IP PBX, e.g. meet-me conference, auto attendant, voice mail, music on hold, etc. Any configuration change to a feature requires clicking **RELOAD** in **Service -> IP PBX Service** to take effect.

5.1 Call Park

During a call, the callee may want to continue the conversation using another phone. The call park feature enables so by letting the callee transfer the call to the call park pilot number. IP PBX will respond an available park line from the pool of call park numbers to the callee. After that the callee may hang up current phone, move to another phone, and dial the park line number told by IP PBX to resume conversation with the caller. If the callee does not call the given park line number to retrieve his call before timeout, IP PBX will ring the original extension where the callee answered the call. To configure Call Park feature, select Feature -> Call Park.

- Enter settings shown in Table 5.1-1.
- Click **APPLY**.

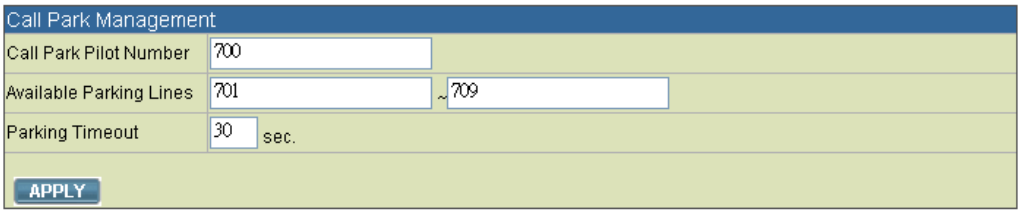


Figure 5-1 Call Park Management

Table 5.1-1 Call Park Configuration Settings

Field	Description
Call Park Pilot Number	A unique extension number for call parking, e.g. 700.
Available Parking Lines	An extension pool for call parking, e.g. 701-720 forms a 20-line pool available for system to park calls.
Parking Timeout	Timeout waiting for picking up the parked call

5.2 Life Line

Life line feature allows specification of emergency number patterns to seize a PSTN line with absolute priority. For example, someone dials an emergency call while all PSTN lines are in use. In such case, if the called number matches any specified life line pattern, the PSTN line with longest talk time so far will be disconnected right away to allow the emergency call.

Select **Feature -> Life line** to configure life-line feature.

5.2.1 Add a Life Line Pattern

- Enter settings shown in Table 5.2-1.
- Click **ADD**.

The newly added pattern should display in the Line Pattern.

5.2.2 Delete a Life Line Pattern

- Select a Line pattern.
- Click DEL.

The deleted pattern shall disappear from the Line Pattern.

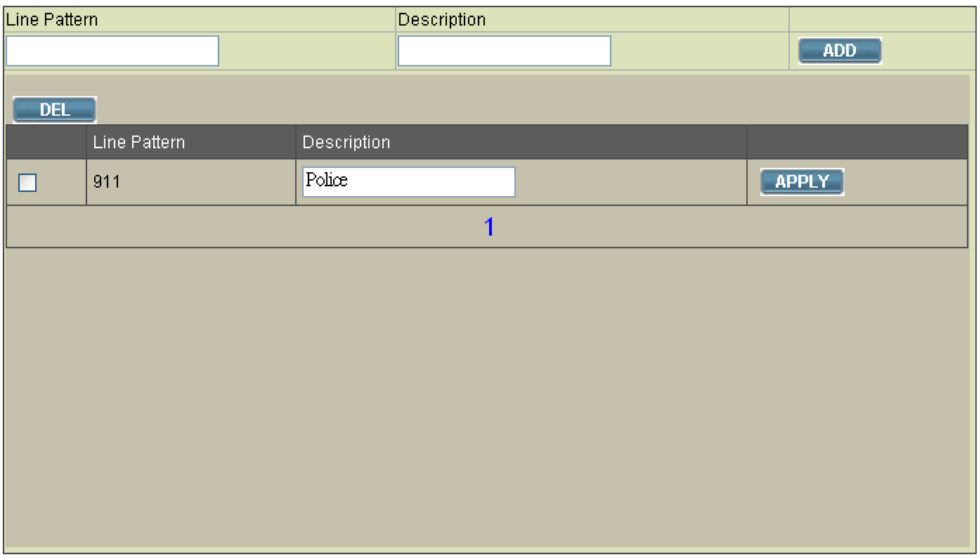


Figure 5-2 Life Line

Table 5.2-1 Life line Configuration Settings

Field	Description
Line Pattern	Pattern for emergency numbers. Note: this is the pattern *after digit stripping*. For example, configure 911 here even if users dial 9911 to reach the 911 service over PSTN when the PSTN trunk has an outbound dialplan of “9.”.
Description	Arbitrary description information.

5.3 Meet-me Conference

Meet-me conference enables conferencing of multiple parties from various sources. A party could dial in a conference from an internal IP phone, an external IP phone on Internet, an analog phone via PSTN, or an IP phone behind another IP PBX. TAINET IPBX 230 allows multiple conference rooms going concurrently using different room numbers. Before entering a meeting room, the caller has to enter the correct PIN of the room number.

Note:

The administrator who invited another meet-me conference room must drop all parties by pressing *5 when the meeting ends.

Select Feature -> Meet-me Conference to configure meet-me conference feature.

5.3.1 Add a Meet-me Conference

- Enter settings shown in Table 5.3-1.
- Click ADD to add a new conference room.

The newly added room should display in the Room Number.

5.3.2 Edit a Meet-me Conference

- Edit settings shown in a row.
- Click APPLY at the end of the row to update the information.

5.3.3 Delete a Meet-me Conference

- Select a room number.
- Click DEL.

The deleted conference room shall disappear from Room Number.

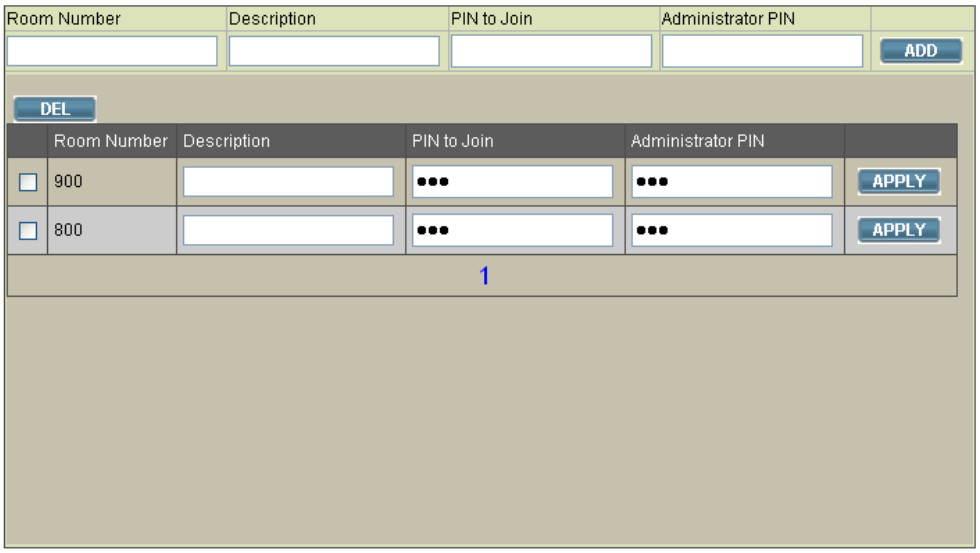


Figure 5-3 Meet-me Conference

Table 5.3-1 Meet-me Conference Configuration Settings

Field	Description
Room Number	Meeting room number, e.g. 8000.
Description	Arbitrary description information.
PIN to Join	PIN for normal users to join the conference. During a conference, a normal user has following options: # to quit conference *1 to mute/unmute *9 to log in as the administrator if there is no administrator dialed in yet.
Administrator PIN	PIN for the administrator of the conference. During a conference, the administrator has following options: # to quit conference *1 to mute/unmute *2 to lock/unlock the conference *3 to invite a user into the conference *4 to drop a party from the conference *5 to drop all parties in the conference *6 to drop the last invited party by *3 ** to send DTMF string to the last invited party by *3. This is useful when the invited party is behind an IVR system.

5.4 Music On Hold

Music-on-hold (MOH) is used in several occasions for a single purpose—to comfort the waiting party with music. One could upload some candidate music files and pick one as the default one.

Select Feature -> Music On Hold to manage MOH files.

5.4.1 Add a MOH File

- Enter settings shown in Table 5.4-1.
- Click ADD.

The newly added file should be displayed in the MOH ID column.

5.4.2 Edit a MOH File

- Edit settings shown as a table at the bottom of the page.
- Click APPLY in the row.

5.4.3 Delete a MOH File

- Select a MOH ID.
- Click DEL at the top-left the table.

The deleted MOH file shall disappear from the MOH ID column.

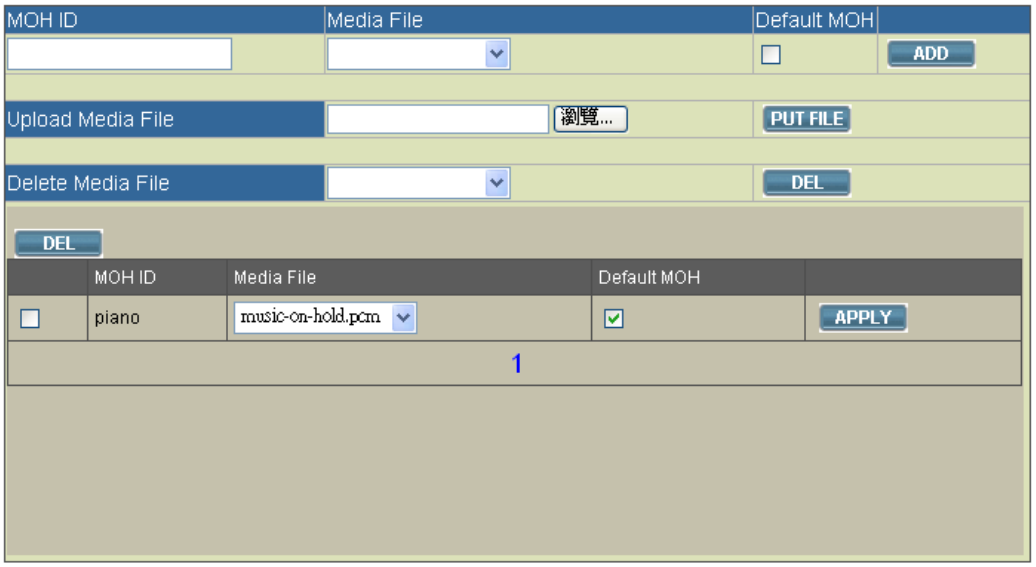


Figure 5-4 Music On Hold

Table 5.4-1 MOH file Configuration Settings

Field	Description
MOH ID	A unique ID containing only alphabets, numbers, and underscore without spaces; 32 characters maximum.
Media File	Candidate music files in the repository. To upload a new music file, click Browse to find a Windows PCM (8000 Hz, 16-bit) file from the local host and click PUT FILE. On successful uploading, the filename will appear in the Media File list. To delete a media file from the list, choose a file from the Delete Media File list, and click DEL to remove it.
Default MOH	Select to use this music file for system default MOH globally.

5.5 Voicemail

TAINET IPBX 230 has a built-in voice mail subsystem with a sophisticated IVR menu. A call to an extension in use or no answer could be configured to enter voice mail recording procedure. After leaving a message, notification e-mail will be sent to the user owns the extension with or without the message in the form of an attached WAV file. The Message Waiting Indicator (MWI) on phone (if any) will be lit. The user could then dial the voicemail pilot number to enter voice mail system to manage messages such as playback, delete, or move them from inbox to different folders. In addition to indicating current voice mail capacity on the management page, IP PBX can send an alarm email to the administrator when the available voice mail space reaches the threshold. To configure Voicemail feature, select Feature -> Voicemail.

- Enter settings shown in Table 5.5-1.
- Click APPLY.

Figure 5-5 Voicemail Prompts Management

Table 5.5-1 Voice Mail Configuration Settings

Field	Description
Voicemail Pilot Number	Number to access voice mail system IVR.
Minimum Message Time	Messages less than this duration will not be notified by e-mail. E.g., 3 (sec).
Maximum Message Time	Maximum duration allowed for a single message. E.g., 60 (sec).
Maximum number of messages per account	Maximum number of messages allowed per extension.
SMTP Server	Hostname or IP address of the SMTP server for voicemail notification.
E-mail from Address	Most SMTP servers require a valid *from* address to accept a mailing request.

Field	Description
Voicemail Available Space Check	Select to enable the Alarm Email function described below.
Send Alarm Email when Space Below	Set a threshold in minutes to send an alarm email to the administrator when the space left is below it.
Voicemail Space Left	Show the available space in Kbytes and minutes. ☞ The storage inside IP PBX saves not only voice mails but also some other stuff, such as CDR and logs. The remained disk space is all for voice mails, and it is the “maximum” available voice mail space.
SMTP Server Account	Specify an account ID if the SMTP server requires authentication for outgoing mails.
SMTP Server Password	Specify the account password if the SMTP server requires authentication for outgoing mails.

Note:

1 Mega bytes memory can store about 2 minutes. And we suggest the format of the USB must be FAT32

5.6 Meet-me Prompts

This page allows replacing built-in meet-me conference prompts with user recordings.

- Click a language and a prompt in the corresponding lists.
- Find a corresponding recording in the local storage.
- Click PUT FILE to complete the replacement.
- To reset a prompt back to default, leave the Upload box in blank and directly click the PUT FILE.

Note that the replacement is done for the selected Language and Prompt only. Currently only following prompts could be replaced.

Figure 5-6 Meet-me Prompt Management

Table 5.6-1 Replaceable Meet-me Prompts

Prompt	Description
Get PIN number	Please enter the conference pin number.
Invalid PIN	That pin is invalid for this conference.
Only Person	You are currently the only person in this conference.

5.7 Voicemail Prompts

This page allows replacing built-in voicemail system prompts with user recordings.

- Click a language and a prompt in the corresponding lists.
- Find a corresponding recording in the local storage.
- Click PUT FILE to complete the replacement.
- To reset a prompt back to default, leave the Upload box in blank and directly click PUT FILE.

Note that the replacement is done for the selected Language and Prompt only. Currently only following prompts could be replaced.

Figure 5-7 Voicemail Prompts Management

Table 5.7-1 Replaceable Voicemail System Prompts

Prompt	Description
Login	Welcome to voice mail system, please enter your mailbox.
Password	Password.
Incorrect Mailbox	Login incorrect, mailbox?
Good-bye	Good-bye.
Prerecording Introduction	Press star (*) to cancel recording and return to the main menu. Or, press pound (#) to start recording right away.
Introduction	Please leave your message after the tone. When done, hang up or press the pound (#) key.

5.8 Broadcast

A User can arrange an event at the exact time in IP PBX. IP PBX will inform all users that set in the Callee Extensions list by ringing their extensions. For example, one arranges a meeting and wants to remind all attendants, he/she may enter settings. When the time set in Date/Time is up, IP PBX will call to the extensions, and then executes the Action to each of the calls. Select Feature -> Broadcast to configure Broadcast feature.

5.8.1 Add a Broadcast

- Click the Add New tab.
- Enter settings shown in Table 5.8-1.
- Click ADD at the bottom of the page.

The newly added broadcast event should display in the Events.

5.8.2 Edit a Broadcast

- Click the Management tab.
- Click an Event.
- Edit settings shown in Table 5.8-1.
- Click UPDATE to update the information.

5.8.3 Delete a Broadcast

- Click the Management tab.
- Select an Event.
- Click DEL.

The deleted broadcast event shall disappear from the Events.

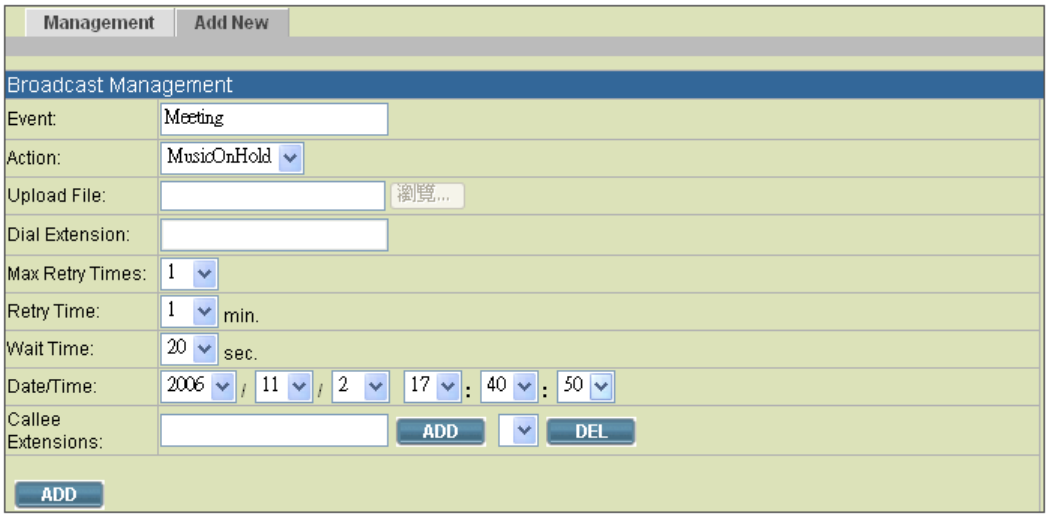


Figure 5-8 Broadcast Management

Table 5.8-1 Broadcast Configuration Settings

Field	Description
Event	A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Action	Select one of the three actions to execute when the Date/Time is up. Playback: Play the uploaded WAV file in the Upload File box to the callee extensions. Dial: IP PBX will ring the party set in the Callee Extensions box first, and then call back to the one in the Dial Extension field to establish a conversation. It is suggested to set one number only in the Callee Extensions unless the extension number in the Dial Extension field is multi-line. MusicOnHold: Play default music to extensions in the Callee Extensions box.
Upload File	Upload a *.wav file, if Playback is selected in the Action list. ☞ The recording format must be 8000 Hz, 16 bits, Windows PCM WAV file.
Dial Extension	Select an extension to call back if Dial is selected as the Action.
Max Retry Times	Maximum redial times if callees did not answer.
Retry Time	A period of time in minutes between two retrying.
Wait Time	Enter timeout in seconds when ringing a callee.

Field	Description
Date/Time	Select a Date/Time to trigger this broadcast event.
Callee Extensions	Intended extensions to be called at the Date/Time.

5.9 Worktime

Worktime defines holidays and business hours for auto attendant or generic IVR application. Several groups of date/time could be defined for different IVR menus. The worktime must collocate with IVR settings .

Select Feature -> Worktime to configure Worktime features.

5.9.1 Add a Worktime

- Click the Add New tab.
- Enter settings shown in Table 5.9-1.
- Click ADD at the bottom of the page.

The newly added worktime should display in the Group ID.

5.9.2 Edit a Worktime

- Click the Management tab.
- Click a Group ID.
- Edit settings shown in Table 5.9-1.
- Click UPDATE to update the information.

5.9.3 Delete a Worktime

- Click the Management tab.
- Select a Group ID.
- Click DEL.

The deleted worktime shall disappear from the Group ID.

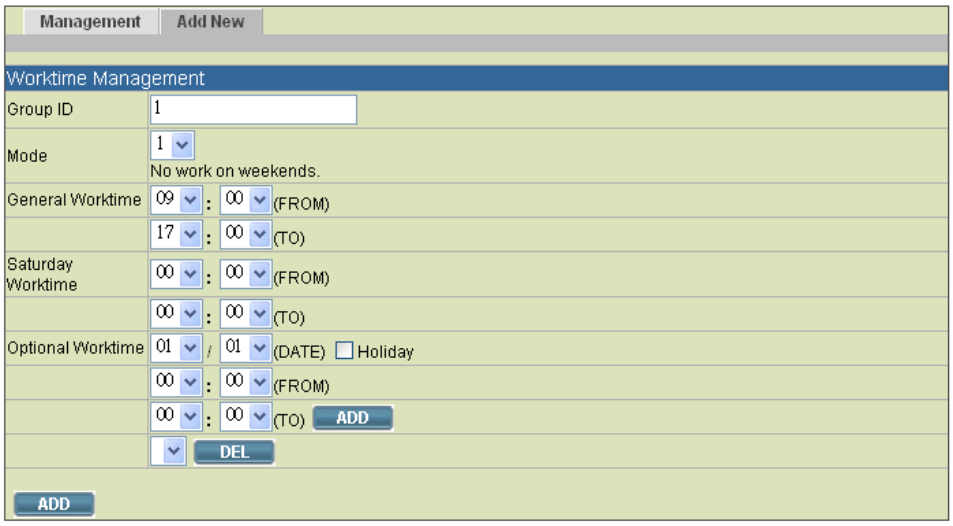


Figure 5-9 Worktime Management

Table 5.9-1 Worktime Configuration Settings

Field	Description
Group ID	A unique ID containing numbers only.
Mode	Select one of the three modes: 1: No work on weekends. 2: Work off and on by turns for weekends 3: Work half-day on Saturdays.
General Worktime	The work time from Monday to Friday.
Saturday Worktime	The work time for Saturdays, this field only active when mode is set to 2 or 3.
Optional Worktime	Special holidays or workday. User can set date and its work time, or set it to a whole-day holiday.

5.10 Memo Call

A user can set a memo at a specific time, e.g. a morning call, in IP PBX to inform the user or another user set in the Extension list by call at the time. When the user picks up the phone, IP PBX plays “voice file” or “music”. The user can also set this memo as a daily route so that IP PBX informs the user at the exact time every day. Select Feature -> Memo Call to configure memo calls.

5.10.1 Add a Memo Call

- Click the Add New tab.
- Enter settings shown in Table 5.10-1.
- Click ADD at the bottom of the page.

The newly added memo call should display in the Memo.

5.10.2 Edit a Memo Call

- Click the Management tab.
- Click a Memo.
- Edit settings shown in Table 5.10-1.
- Click UPDATE to update the information.

5.10.3 Delete Memo Call

- Click the Management tab.
- Select a Memo.
- Click DEL.

The deleted memo call shall disappear from the Memo.

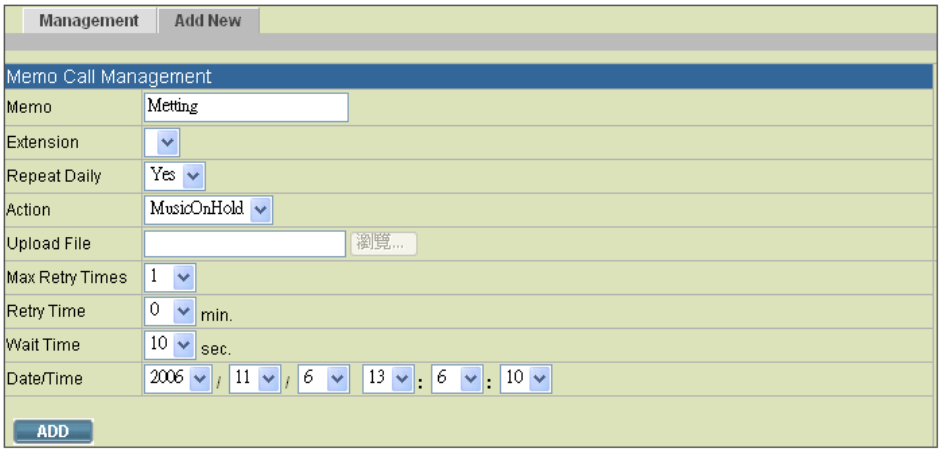


Figure 5-10 Memo Call Managment

Table 5.10-1 Memo Configuration Settings

Field	Description
Memo	A unique ID containing alphabets, numbers, and underscore only without spaces.
Extension	Click an extension to call.
Repeat Daily	Select Yes to enable looping playback every day at the exact time.
Action	Select one out of the two available actions for the memo. Playback: Play the uploaded WAV file in the Upload File box to the extension(s). MusicOnHold: Play default music to the extension(s).
Upload File	Upload a *.wav file, if Playback is selected in the Action list. ☞ The recording format must be 8000 Hz, 16 bits, Windows PCM WAV file.
Max Retry Times	Maximum redial times if callees did not answer.
Retry Time	A period of time in minutes between two retries.
Wait Time	Enter timeout in seconds when ringing a callee.
Date/Time	Select a Date/Time to trigger this memo call.

5.11 Automatic Call Distribution

TAINET IPBX 230 implements Automatic Call Distribution (ACD) by queues and agents. A pool of agents is created first and then one or more queues could be added with distinct service numbers and a list of agent extensions. Each agent should be bound with an existed extension to take calls and each queue could have a specific distribution policy, such as round robin, ring all, least recent, fewest calls, and random. With such settings one can enable agents located at any location to serve a unified queue of calls. This comes with a benefit that administrative changes can be made in a single place, and these changes immediately become effective for all calls and all agents. This not only enables distributed centers to be operated cost-effectively, but also reduces the cost and complexity of implementing home-based agents, who now only need an IP connection to serve calls in queues. Refer to Table 5.11.1 for creating an agent pool, and Table 5.11.2 for setting queues. Select Feature -> ACD to configure agents and queues.

5.11.1 Set Agent Login and Logout

Before add agents or queues of the ACD, the administrator has to give universal Agent Login and Logout numbers for all agents.

- Click the Agent Login / Logout tab.
- Enter numbers in Agent Login and Agent Logout.
- Click MOD to apply login and logout numbers to all agents.

5.11.2 Add an Agent or a Queue

- Click the Add New tab.
- Enter settings shown in Table 5.11.1 and Table 5.11.2.
- Click ADD at the bottom of the page.

The newly added agent or queue should display in the Management page.

5.11.3 Edit an Agent or a Queue

- Click the Management tab.
- Click a queue or an agent in Queue or Agents.
- Edit settings shown in Table 5.11.1 for the queue or Table 5.11.2 for the agent.
- Click UPDATE to update the information.

5.11.4 Delete an Agent or a Queue

- Click the Management tab.
- Click a bottom of the row.
- Click DEL.

The deleted agent or queue shall disappear from the Management page.

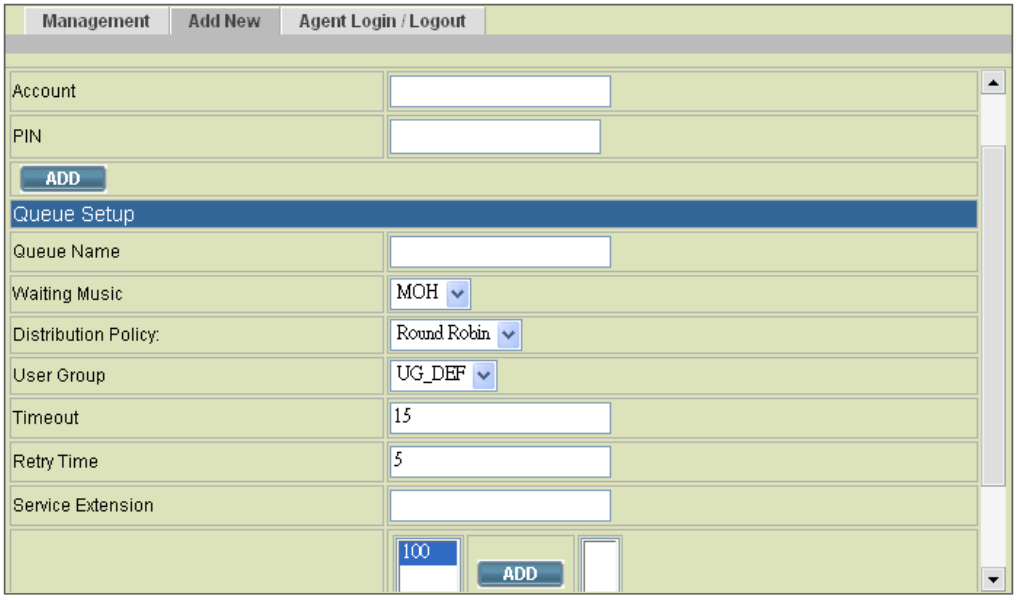


Figure 5-11 Add new settings

Table 5.11.1 Agent Configuration Settings

Field	Description
Agent Name	Name the agent. A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.
Account	Enter an account number for the agent.
PIN	Enter a login password for the agent.

Table 5.11.2 Queue Configuration Settings

Field	Description
Queue Name	Name a queue. A unique ID containing alphabets, numbers, and underscore only without spaces; 32 characters maximum.

Field	Description
Waiting Music	The music to be played before any agent answers the call.
Distribution Policy	Ways of distributing calls to the agents in a queue: Round Robin: take turns to ring each available agent. Ring All: ring all available agents until one answer (default). Least Recent: ring the agent that was least recently called in this queue. Fewest Calls: ring the agent with fewest completed calls in this queue. Random: ring the agents randomly.
User Group	Select a usergroup for this queue.
Timeout	Set a timeout period in seconds when no agent answers the call. ☞ The time must be longer than Timeout To Next Forward of the extension, or the call will forward to an extension that is set in Unavailable Call Forward in Extension of IP Phone.
Retry Time	Set a waiting time in seconds to ring agents in the queue again if the call is not answered.
Service Extension	Set the representative number of this queue, e.g. a queue named Support, and its service extension is 500. There are three agents with extensions of 155, 157, and 166 in Support. The Distribution Policy is set to Ring All. When a customer calls the service extension 500 for Support, extensions 155, 157 and 166 will all ring simultaneously.
Agent List	Click ADD/DEL to select/remove agents belonged to this queue. The left box lists the available agents, and the right box shows the selected agents.

5.12 Interactive Voice Response (IVR)

Interactive Voice Response (IVR) helps a caller to select options from voice menus by pressing keys on a telephone keypad. With IVR, a caller can connect to an expected extension or a service promptly. TAINET IPBX 230 enables multiple configurable IVR menus in a single system, and each of them could have a hierarchy up to three layers. Select Feature -> IVR to add, edit and delete the IVR menus. You can also manage IVR prompts, used by IVR menus, in this page.

5.12.1 Add a new IVR Menu

- Enter a name of an IVR menu in IVR Name, and click a file in the Prompt list.
- Click ADD next to the IVR Name box to set the new IVR name in Info. System will prompt to ask for confirmation whether a Worktime setting is required or not. This is because Worktime setting can only be added when creating a new IVR. After creation, an IVR without Worktime setting cannot be associated with a Worktime setting later. If Worktime setting is indeed not required, click Cancel in the popped dialog.
- Enter settings shown in Table 5.12.
- Click APPLY to add the new IVR menu and see it as a tree view in Info.
- For example, to create a basic Auto Attendant IVR for a trunk with Usergroup of Privilege dial_in:
  - Specify an **IVR Name**, say *Basic_AA*
  - Choose **/agent-newlocation.gsm* from **Prompt** list in **Action Data** block.
  - Choose *dial_in* from **Group** in **Action Data** block.
  - Click the **ADD** button beside the given IVR Name, *Basic_AA*.
  - Click **Cancel** to the popped dialog window to confirm Worktime setting is not required.
  - Now, *Basic_AA* should be available in the trunk configuration page for IVR setting.

5.12.2 Edit an IVR Menu

- Click an IVR name in the All IVR Menus list.
- Edit settings shown in Table 5.12.
- Click APPLY to update the changes.

5.12.3 Delete an IVR Menu

- Click an IVR name in the All IVR Menus list.
- Click DEL to delete the IVR menu.

Table 5.12 Interactive Voice Response Configuration Settings

Field	Description	
All IVR Menus	Select a preferred IVR menu name.	
Info	View the IVR menu as a tree view.	
IVR Name	Specify the name of the IVR.	
Rule	Click a number in the Keypad list and one of the following actions in the Action list to associate an action with a key.	
	Hang Up	To cut off the call immediately.
	Play Back	To play the IVR prompt selected in Prompt list
	Call To	To call an extension.
	Go to Top	To go back to the root menu of the IVR.
	Next Layer	To go to the next layer of the IVR menu.
	Select Language	To choose a language.
	Return	To go back to the previous layer.
Node	Information of the configured keys and actions. Click a node and DEL to delete the node and its underlying structure.	
Child Rule	If a Next Layer is selected, Child Rule sets the key-action associations with the next-layer menu.	
Action Data	Specify applicable parameter(s) for an action.	

Field	Description	
	Prompt	Select a *.wav recording file that you add from the IVR Prompt tab, or select one of the following default voice file. The default file that marked */ in front of the file name means this voice file provides all languages that IP PBX has for you to select. Please click a language in the Languages list. agent-newlocation.gsm: Please enter a new extension followed by the # key. auth-thankyou.gsm: Thank you. invalid.gsm: I'm sorry, that is not a valid extension, please try again. transfer.gsm: Please hold while I try out that extension. ss-busy.gsm: System is busy at this moment, please try again later. ss-noservice.gsm: The number you have dial is not in service, please check the number and try again. vm-goodbye.gsm: Good bye. vm-sorry.gsm: I'm sorry, I do not understand you response.
	Group	Select a usergroup.
	Language	Select a language of the IVR.
	Extension	Enter an extension number to be transferred to.
Active Worktime	Select to set work time for the IVR.	
Group	Select a work time group set in Feature -> Worktime.	
In-Hour Actions	Select one action during business hours.	
	Play Back	To play the selected prompt.
	Call To	To transfer to an extension
	No Action	No action.
Prompt	Select a *.wav file if Playback is selected in the In-Hour Actions list.	
Extension	Enter an extension number if Call To is selected in the In-Hour Actions list.	

Field	Description	
Off-Hour Actions	Select one action during the off hours.	
	Play Back	To play the selected prompt.
	Call To	To transfer to an extension
	No Action	No action.
Prompt	Select a *.wav file if Playback is selected in the Off-Hour Actions list.	
Extension	Enter an extension number if Call To is selected in the Off-Hour Actions list.	

5.12.4 IVR Prompts Management

One can upload customized IVR prompts in Feature -> IVR, and click IVR Prompts Management tab.

5.12.4.1 Add an IVR Prompt

- Select a language from the Language list.
- Click Browse to find the expected recording in the local storage.
- Click PUT FILE to upload the file add it to the Prompt list.

5.12.4.2 Delete an IVR Prompt

- Select a *.wav file from the All Files list.
- Click DEL.

The deleted file shall disappear from the All Files list.

Chapter 6. Example Provisioning

This chapter introduces several practical configuration examples of TAINET IPBX 230 deployment. The configuration of TAINET IPBX 230 is very flexible and the expressiveness of usergroups, routegroups and trunks are scalable enough to support various network architectures. Users could refer to these examples and build a larger network involving multiple sites and advanced services.

6.1 Internal Extension Configuration

The procedure introduced below is recommended as the first step of a configuration task. The configuration only enables internal extension calls, but it serves as a good practice for administration. User can fllow the steps asFigure 6-1.

- Configure a usergroup named ALL (refer to 4.2).
- Configure users and assign every user to usergroup ALL (refer to 4.1).
- Configure devices and an extension for each device (refer to 4.3).
- Assign each extension to a corresponding user (refer to 4.3).
- Configure each client phone with respect to the extension number and password in its IP PBX extension configuration accordingly.
- Reload the IP PBX service (refer to 3.5.2).

Up to this point all configured phones should register with the IP PBX with a usable extension. Since these phones are all belonged to the same usergroup ALL, they can call one another without limitation.

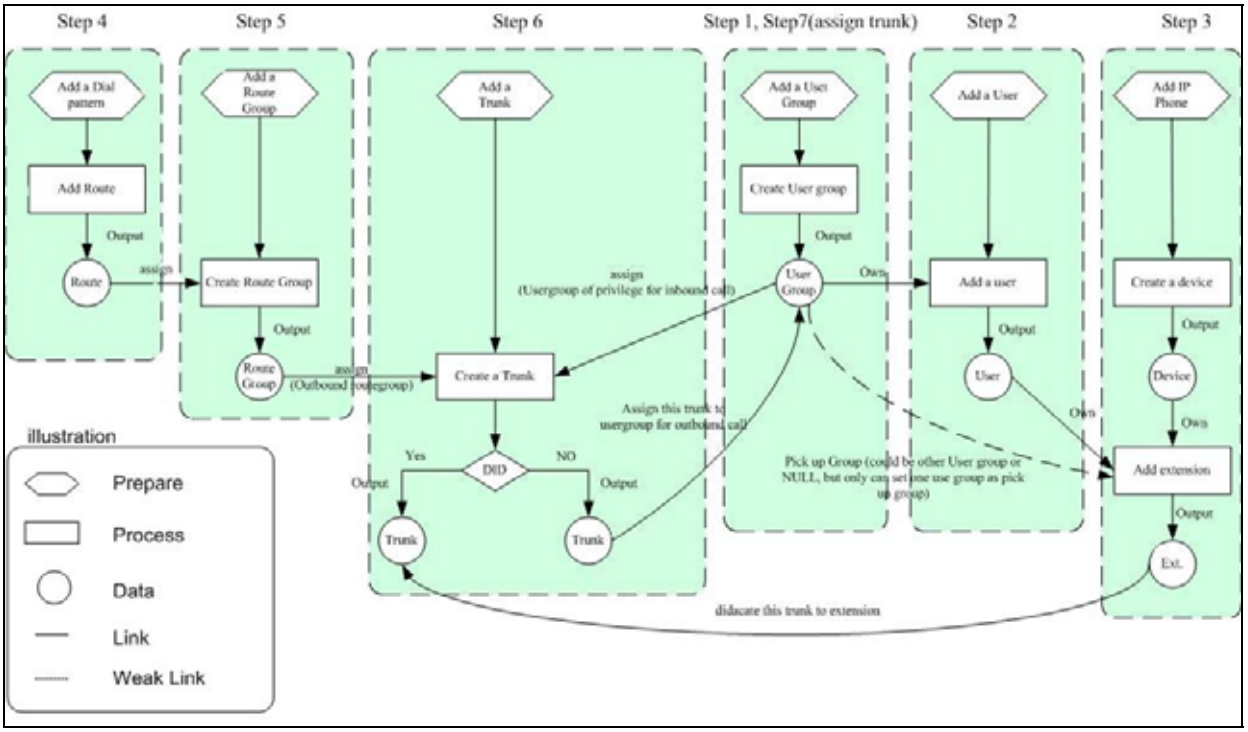


Figure 6-1 Settings Roadmap

6.2 Case I: Single-site Configuration

This case describes the typical settings of a single-site configuration; say Company A. Assume Company A has a DSL connection for Internet access and 2 PSTN subscriber lines as shown in Figure 6-1. The provisioning tasks include:

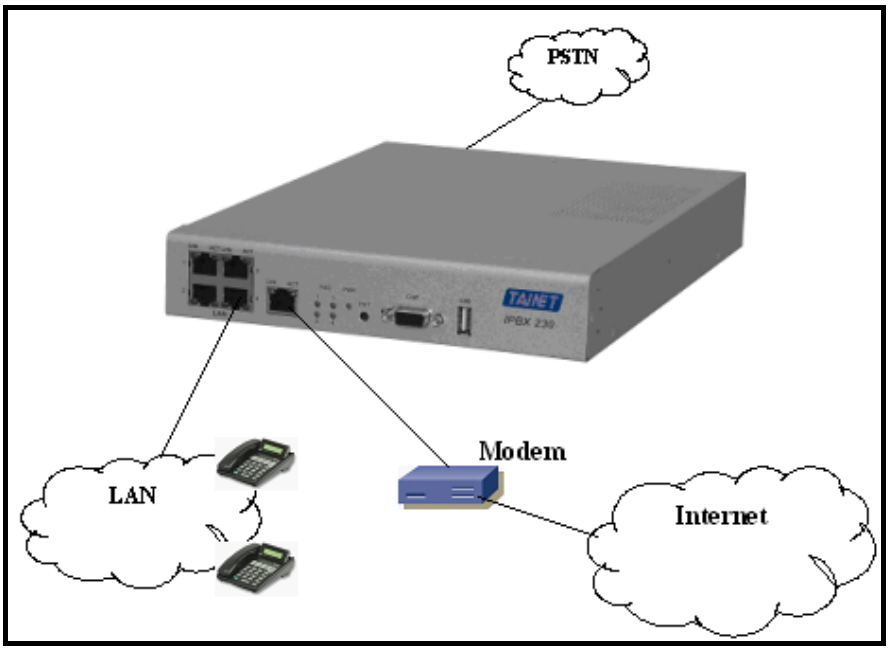


Figure 6-2 Single side Setup topology

- There are staff phones in cubes and offices, and utility phones in public areas.
- Each phone has one extension, and can call any extension without limitation.
- Only staff phones can call out to PSTN with a prefix 9.
- Incoming PSTN calls are answered by auto attendant and could be transferred to any extension.

Configuration steps:

- Create usergroups named staff, utility, and ext-all.
- Add staff and utility in the Reachable User Groups of ext-all.
- Create a user account for each staff and assign it to usergroup staff.
- Create an additional user account named public and assign it to usergroup utility.
- Create a device for each physical phone and designate an extension.

- Assign extensions of staff phones to corresponding users.
- Assign all extensions of utility phones to share the same user public.
- Create a route, named pstn with pattern “9.” number of digits stripped “1”, and no prefix.
- Create a routegroup named pstn-out, and add route pstn only.
- Create a PSTN trunk with ID “1”, port “1-2”, choose pstn-out as outbound routegroup, do not select any DID of extension, and select ext-all as the usergroup of privilege.
- Return to usergroup configuration. For usergroup staff, click pstn1 in Associated Trunks, and select utility in Reachable User Group; while for usergroup utility, only select staff in Reachable User Group.
- Reload the IP PBX Service.

6.3 Case II: Two-site configuration

This case describes the typical settings of a two-site configuration; say Company B headquarters B-HQ and its branch B-BR located in another country. Assume each site has a DSL connection for Internet access. B-HQ has 4 PSTN subscriber lines and B-BR has 2 lines as shown in Figure 6-2. The provisioning tasks include:

- Both sites have staff phones in cubes and offices and utility phones in public areas.
- Each phone has one extension. A utility phones can call extensions within the site it is in only, while the staff phones can call any extension in both sites without limitation. B-HQ has extensions 1XX and B-BR has extensions 2XX.
- Calls between B-HQ and B-BR use private SIP trunks across Internet. IP PBX at B-HQ has a static IP address, 64.1.0.1 and IP PBX at B-BR has a static IP address, 222.44.0.1.
- Only staff phones can call out to PSTN with a prefix 9.
- B-HQ staff phones call out starting with 90118621 will relay to B-BR through the SIP trunk and then hop off to PSTN in B-BR. Similarly, B-BR staff phones call out starting with 9001408 will relay to B-HQ for PSTN hop-off.

Incoming PSTN calls are answered by auto attendant and could be transferred to any extension.

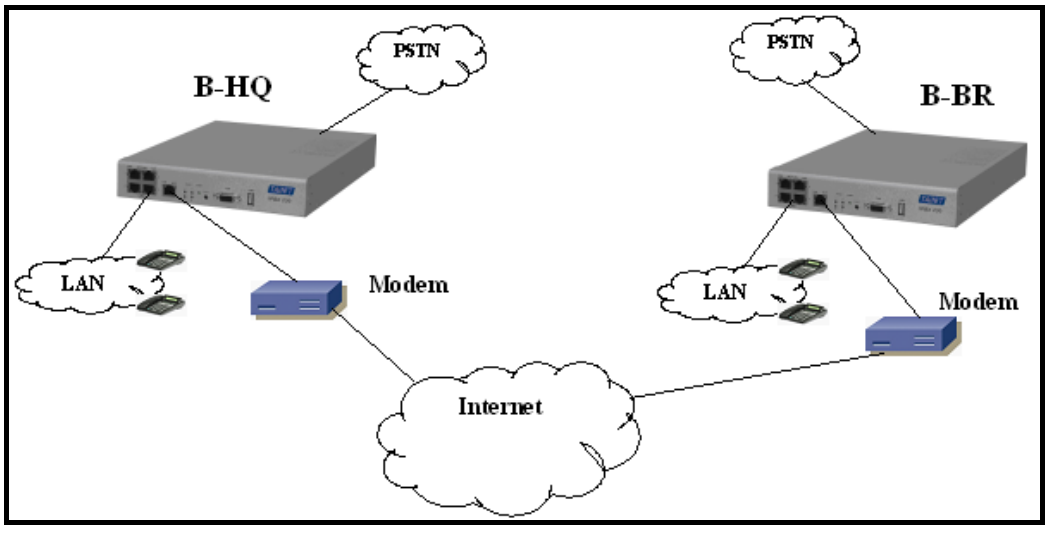


Figure 6-3 Two Side Setup topology

Configuration steps in B-HQ:

- Create usergroups named staff, utility, and ext-all.
- Add staff and utility in the Reachable User Groups of ext-all.
- Create a user account for each staff and assign it to usergroup staff.
- Create an additional user account named public and assign it to usergroup utility.
- Create a device for each physical phone and designate an extension.
- Assign extensions of staff phones to corresponding users.
- Assign all extensions of utility phones to share the same user, public.
- Create a route, pstn, with pattern “9Z.” with number of digits stripped “1”, no prefix.
- Create a route, pstn-br, with pattern “90118621.” with number of digits stripped “8”, prefix “9”.
- Create a route, ext-br, with pattern “2XX” and number of digits stripped “0”, no prefix.
- Create a routegroup, pstn-out, containing route pstn only.
- Create a routegroup, to-br, containing routes pstn-br and ext-br.
- Create a PSTN trunk with ID “1”, port “1-4”, choose pstn-out as outbound routegroup, do not select any DID of extension, and select ext-all as the usergroup of privilege.
- Create a dynamic peer SIP trunk with ID 100; password hq-secret; choose to-br as outbound routegroup, do not select DID, and select staff as the usergroup of privilege.
- Return to usergroup configuration. For usergroup staff, select 100, pstnl in Associated Trunks, and select utility in Reachable User Group; while for usergroup utility, only select staff in Reachable User Group.

Reload the IP PBX Service.

Configuration steps in B-BR:

- Create usergroups named staff, utility, and ext-all.
- Add staff and utility in the Reachable User Groups of ext-all.
- Create a user account for each staff and assign it to usergroup staff.

- Create an additional user account named public and assign it to usergroup utility.
- Create a device for each physical phone and designate an extension.
- Assign extensions of staff phones to corresponding users.
- Assign all extensions of utility phones to share the same user, public.
- Create a route, pstn, with pattern “9Z.” with number of digits stripped “1”, no prefix.
- Create a route, pstn-hq, with pattern “9001408.” with number of digits stripped “7”, prefix “9”.
- Create a route, ext-hq, with pattern “1XX” and number of digits stripped “0”, no prefix.
- Create a routegroup, pstn-out, containing route pstn only.
- Create a routegroup, to-hq, containing routes pstn-hq and ext-hq.
- Create a PSTN trunk with ID 1, port 1-2, choose pstn-out as outbound routegroup, do not select any DID of extension, and select ext-all as the usergroup of privilege.
- Create a SIP trunk with ID 100 pointing to 64.1.0.1 port 5060; password hq-secret; choose to-hq as outbound routegroup, do not select DID, and select staff as the usergroup of privilege.
- Return to usergroup configuration. For usergroup staff, select 100, pstnl in Associated Trunks and select utility in Reachable User Group; while for usergroup utility, only select staff in Reachable User Group.

Reload the IP PBX Service.